Legal Advice for Songwriters

# Electronic Musician

JUNE 2007

PERSONAL STUDIO | RECORDING | PRODUCTION | SOUND DESIGN

## 11 POWERHOUSE SOFTWARE SAMPLERS THAT DELIVER SONIC MAGIC

## DISCOVER THE LOGIC OF APPLE LOGIC PRO

## HOW TO OPTIMIZE LEVELS IN YOUR DAW

## REVIEWS

ADAM A7
Universal UAD-1e Expert Pak
M-Audio NRV10
Akai Pro EWI4000s
*and 5 more*

## HOLLYWOOD SOUND DESIGNER Scott Gershin

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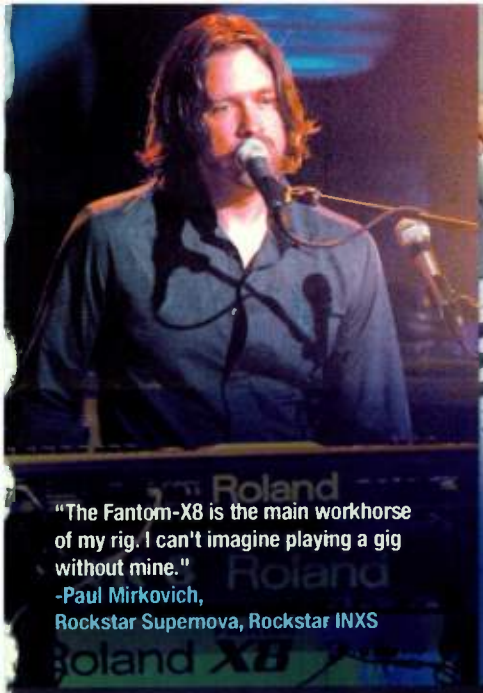

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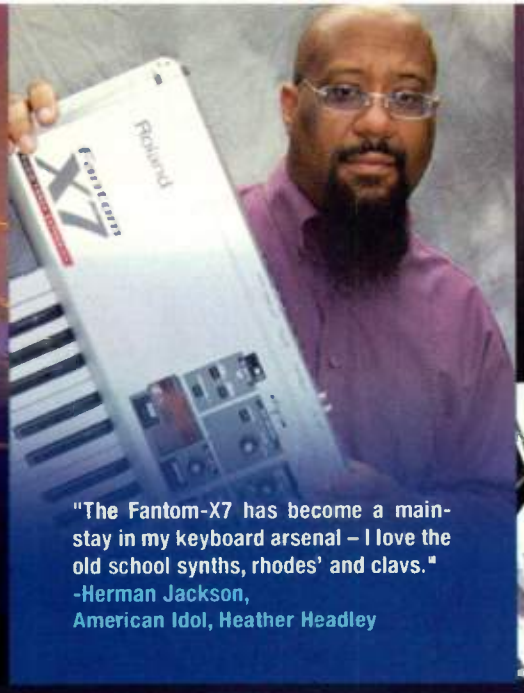

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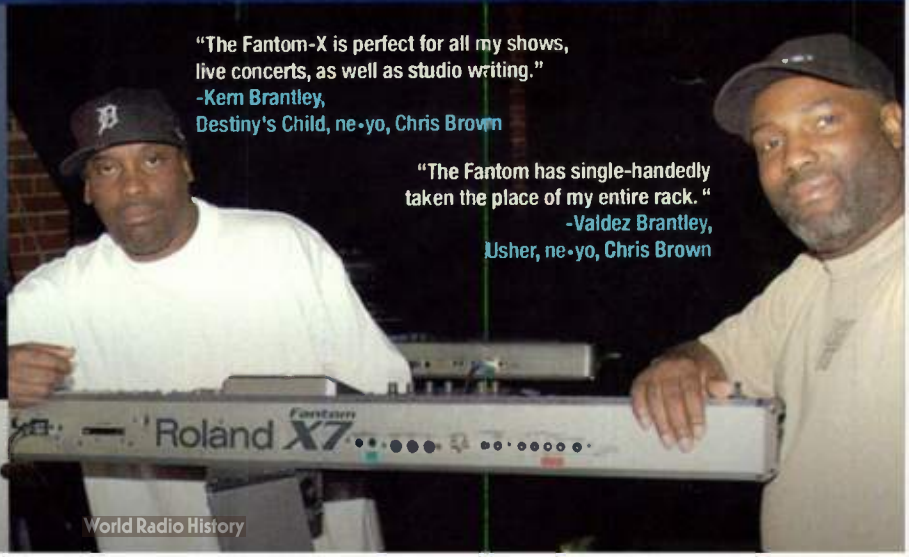

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**2006** AudioSnap, Active Controller Technology introduced; 1st to support VST 2.4

**2005** 1st DAW with end-to-end 64-bit audio, Windows x64 support; BitBridge introduced to run 32-bit VSTs on x64; Cakewalk Instruments launched

**2004** SurroundBridge introduced for using stereo FX in surround

**2003** 1st DAW with advanced multi-processor support, Universal Bussing Architecture introduced

**2002** MIDI Groove Clips introduced, 1st DAW to support both ASIO & WDM

**2001** SONAR introduced: 1st DAW to combine MIDI & audio, ACID-style looping, & virtual instruments

**1999** WavePipe technology for low latency audio streaming

**1998** 1st DAW with synchronized host-based playback of MIDI, audio, & video; MIDI FX introduced

**1997** 1st native DAW for Windows NT, 1st with real-time DirectX FX; StudioWare introduced

**1995** Cakewalk Pro Audio: 1st native 32-bit MIDI & digital audio workstation for Windows 95

**1993** Real-time MIDI editing introduced

**1991** Cakewalk Professional for Windows: 1st sequencer for Windows 3.1, CAL (Cakewalk Application Language)

**1987** Cakewalk for DOS introduced, 256 tracks



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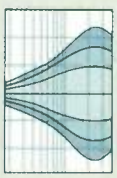

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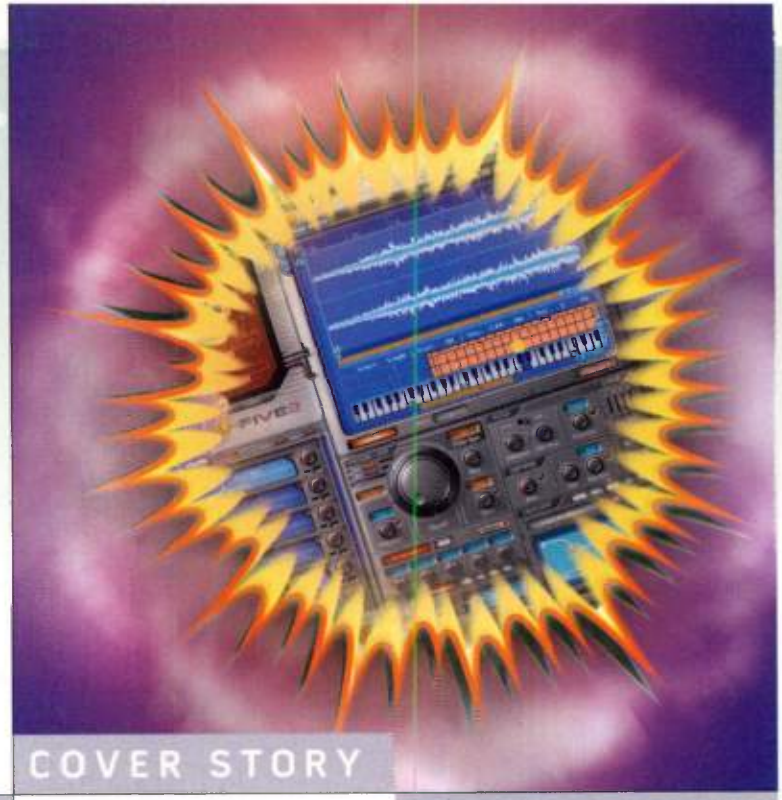
# INSIDE



COVER STORY

DMITRY PANICH

## FEATURES


### 31 MASTER CLASS: THINK DIFFERENT

The Apple Logic Pro 7.2 digital audio sequencer has incredible power, but learning to use its features can be extremely challenging. We reveal the logic of Logic Pro and show you how to get past some of its stumbling blocks and take advantage of its distinctive design. By Len Sasso

### 40 POWERHOUSE SAMPLERS

The incredible feature sets found in modern software samplers are so superior to those in hardware samplers that the latter have become all but obsolete. EM compares and contrasts 11 top soft samplers to help you choose the right one for your needs. We also provide an exclusive advance look at Digidesign Structure and MOTU MachFive 2. By Geary Yelton

### 63 PRODUCTION VALUES: IT'S ONLY MAKE-BELIEVE

Acclaimed Hollywood movie and video-game sound designer Scott Gershin talks about his work, the techniques he uses to construct sounds, the enormous sonic potential in normal household sounds, and more. By Mike Levine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## DEPARTMENTS


10 FIRST TAKE

14 LETTERS

18 EMUSICIAN.COM TABLE OF CONTENTS

20 WHAT'S NEW

112 MARKETPLACE

115 CLASSIFIEDS

Electronic Musician
June 2007

122

# INSIDE

## COLUMNS

28 TECH PAGE **Thanks for the Memories**
Molecular memory could maintain Moore's law well beyond silicon's limits.

70 MAKING TRACKS **Max Headroom**
The rules when gain staging digital signals are different than with analog.

74 SOUND DESIGN WORKSHOP **A Different Drummer**
Vocoding with drum sounds instead of speech.

76 SQUARE ONE **Preaching to the Converted**
Learn the magic of A/D/A conversion.

80 MUSIC BUSINESS INSIDER **Q&A: Bev Green**
An attorney offers sound legal advice for composers and songwriters.

122 PRO/FILE **Tracking in Soulville**
Recording the blues of the Soul of John Black.

84

## REVIEWS

84 **ADAM PROFESSIONAL AUDIO A7** powered monitor

88 **UNIVERSAL AUDIO UAD-1e Expert Pak** (Mac/Win) DSP card and plug-ins

94 **M-AUDIO NRV10** analog mixer with FireWire interface

98 **AKAI PROFESSIONAL EWI4000s** MIDI wind controller

104

## 104 QUICK PICKS

**FXpansion *BFD Percussion*** sample library

**Echo Digital Audio AudioFire4** (Mac/Win) FireWire audio interface

**WaveMachine Labs Drumagog Pro 4** (Mac/Win) drum replacement effects plug-in

**Karma Audio K6** ribbon microphone

**Line 6 LowDown Studio 110** bass amplifier

# A **Symphony** that Rocks... Hard.


## Ulrich **Wild**

Producer, Engineer, Mixer

"Technology has finally caught up to my needs in a DAW. I'm running Logic Pro with the new Symphony System with fantastic results. The Symphony System finally brings near zero latency to native recording, and the new converters (AD/DA-16Xs) sound absolutely fantastic. ***Logic Pro sounds much better than any other DAW I've used before.*** The low-end is warm and the high-end is silky smooth. It is a very efficient program and not very taxing on the CPU, which, along with the new Intel Macs, leaves me with an astounding surplus of processing power. ***The quality of this system meets and even surpasses the large format analog consoles I was used to.*** I have mixed many Rock and Metal records and until recently, I wouldn't even have considered mixing "in the box". Now, however, I mix nearly all projects in Logic."

**Recent projects:**
Project 86, SOiL, Static X, DethKlok, Mindless Self Indulgence, Breaking Benjamin

## The **Symphony** System:

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See Ulrich's system and more Symphony users at:
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# Of Death, Taxes, and Roundups

Benjamin Franklin famously wrote, "'In this world, nothing is certain but death and taxes." He was wrong, of course. For one thing, some people legally avoid paying taxes. For another, anytime EM does a product roundup, especially about software, at least one major company is certain to inform us that it is going into the beta-testing phase with an important new version of a product we have included in our article that "changes everything." And, of course, that product will ship by the time the magazine is in print but not soon enough for us to test the final version for our roundup. This usually happens after we are already committed to publishing the story as scheduled. Trust me, your chances of avoiding taxes are far better than are our chances of avoiding this problem.

So it was no surprise that just when Associate Editor Geary Yelton was rolling along with his research for this issue's cover story (see "Powerhouse Samplers" on p. 40), we discovered that two major products were entering beta testing and both would be shipping at or near our newsstand date. One is a new product, Digidesign Structure; the other is a major rewrite of a program, version 2 of MOTU's MachFive. Naturally, without these two new products, our roundup would be closer to a bum steer.

JULIE BIRMINGHAM

While I disagree with Franklin's principle of certainty, I heartily agree with the old proverb "If life deals you lemons, make lemonade." And indeed, we were ultimately able to turn a first-rate problem into a first-rate article.

The big breakthrough came when, after careful negotiations, both Digidesign and MOTU agreed to give us early beta copies of their new software samplers. We had to wait until the companies were confident that the betas were reasonably stable, and Geary surely held his breath when he first installed and launched them. Fortunately, his computer did not blow up, and he was able to test the two new software samplers thoroughly and include them in our story. Instead of a roundup that would be dated the day we published it, we have a fine story that, to my knowledge, features the first-ever hands-on evaluations of Structure and MachFive 2.

This does not alter our long-standing policy against evaluating beta software in our Reviews section. Betas are not finished software, and although most, if not all, of their features are usually functional, they may be unstable and the developers may still be making decisions about the final feature set. In a regular product review, we want to test the same version that our readers would buy.

But I'm glad that we included the two betas in "Powerhouse Samplers." Our story states clearly that we did not test the shipping version, and we did not criticize the software for the sorts of problems that betas typically have. But we were able to discuss the functionality and usefulness of the main feature set and user interface, as well as the general gestalt of the program.

We appreciate the trust that Digidesign and MOTU showed in giving us early beta software so we could complete this important cover story. Thanks to their cooperation and Yelton's hard work, we have served up some very tasty, refreshing lemonade. Try some!

**Steve Oppenheimer**
**Editor in Chief**


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LAURA WILLIAMS

# Letters

## Lassoing Laskow

In response to the interview with Michael Laskow, I would like to point out a few things (see "Music Business Insider: Q&A: Michael Laskow" in the March 2007 issue of EM). On p. 76, Laskow says, "In a perfect world, people could go to a Web page one time and fill out a survey . . ." Well, Pandora (www.pandora.com) does just that.

Also, one glaring point I think he failed to mention is how businesses are constantly being robbed by P2P systems and the Internet. This will never stop until a system is put in place to monitor these illegal activities. The ISPs should be forced to hand over names and place filters to monitor and track illegal activities. Until that happens, we stand to lose way more than we could ever have expected.

We need people like Laskow to spearhead the movement that downloading is illegal. Rather than just hearing about the RIAA suing a few individuals, we should see ad campaigns and so forth that put the issue in the forefront. Otherwise, many jobs will be lost, movies and music will fade away, and we will be left with a huge void in our entertainment industry.

**Kevin Jardine**
**via email**

On p. 78 of the Michael Laskow interview, Laskow says, "You don't need to go to a $300-per-hour, 48-track digital studio . . ." He's absolutely right. People can go to somebody's basement and have some hack engineer tear their recording apart and make it sound like utter crap. Laskow is supposed to have experience in A&R? How many horrible recordings has he tossed because of the sheer disgusting quality of the demos?

I've produced many bands at multimillion-dollar recording studios, and band members always say they work better in a big-studio atmosphere. I thought this was common knowledge. Big-room studios with their 15-foot-long SSL consoles are on the way back, buddy. Leave the hack engineers with their $200 8-track boards and their "in-the-box" rigs to the bands that don't have enough heart or drive to come up with a few thousand to get into a real studio.

**Nik Tyler**
**Philadelphia, Pennsylvania**

## More for Modartt

I must respond to the Modartt Pianoteq review that was, well, pathetic (see the March 2007 issue of EM). It was basically a restatement of the manual and the pop-up tool tips, and the bit about player and listener position was so inane that I almost fell off the toilet when I read it.

The entire concept of modeled pianos seems to be escaping someone here, as well as what constitutes depth in a product review. No, there's not a given set of presets that produce the any-brand piano sound—what is given is the ability to build your own freaking piano, which is not only brilliant on Modartt's part, but also a true gift to the rest of us.

Modartt's Damper Up and Key Up resonance model is astonishingly good. This feature alone is probably the most important sonic element in a piano model, the very thing that adds depth and harmonic character to notes, chords, and expressive damper pedal use. It got one line in the review. True modeled resonance accomplishes with mathematics what happens when a "real" piano is played, which is to define what harmonic content is available for any given note played based on what else is not damped and available. The reality of sampled pianos is that they are no more or less than audio clip art. On a sample, the sound is always a particular note, at a particular point in time, relative only to itself and the recording variables. Forget the number of Velocity layers and convolution paste-ins. It is still an audio picture of a moment in time, every time.

It is unfortunate that none of us playing a piano will ever know what it sounds like in section GG, row 9, seat 14. So, where in the audience would Mr. Sasso like to be sitting, and would he agree with the preset? Why not apply a decent (probably virtual) ambience-generating appliance and make your own decision? Does an any-brand "real" piano come with this feature?

Considering the price of piano sample libraries and the hundreds of dollars required to play just one piano, the paltry $325 for all the pianos in Pianoteq is a steal, not a 3 out of 5 for value.

**Phil Huston**
**via email**

*Phil—Thanks for your detailed analysis, and sorry to hear about your near*

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### COVER

**Correctional Facilities: 12 Ways to Fix a Bad Mix**

Learn how to avoid or correct typical problems such as spectral- and track-volume imbalances, poor dynamics, and collapsed stereo image.

### FEATURES

**Streamlining the Project Studio**

Plan your personal-studio setup for maximum efficiency and ergonomic convenience. We examine considerations such as cabling, acoustic treatment, and the placement of monitors, racks, mixers, keyboards, and computers.

**Production Values: Bill Bottrell**

Bill Bottrell has been at the heart of classic albums by artists such as Sheryl Crow, Michael Jackson, and Rosanne Cash. In this interview, he talks about his technical and aesthetic approach to production, including how he coaxes the best performances from his artists.

### COLUMNS

**Making Tracks: ReCycling in Ableton Live**

Slice and dice audio files using MIDI trigger files in Live 6.

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After slicing and dicing in Live 6, use its Impulse sample player to play your ReCycle-style slices.

**Square One: Take Your Music and Stuff It**

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**. . . and much more**

# Letters

*accident. I think a rereading of the review, which was highly regarded by the manufacturer, might lead to a reassessment. To be clear, I consider Pianoteq a seminal achievement and said so in the review.*

*I did mention listening perspective. High-end virtual pianos usually allow you to switch channels, thus offering a player's or listener's perspective. It's not a big issue, but it's a valid point of comparison. I did point out that there are no name-brand piano presets, because that is, for good or ill, expected in a virtual piano. However, I went on to effuse about the many things you get with a physical-modeled instrument that you will never get from samples. In fact, I spent most of the review on those features. I did discuss room ambience, because that is an important distinction between sampled and modeled instruments. Far from missing the point, I would be derelict to leave it out. However, I went on to say that a good convolution reverb would solve the problem, if indeed the reader hears it as a problem. Finally, at EM, a 3 for value means "good; meets expectations." For a $325 virtual instrument, no matter what it is, that is a good score, and I stick by it. —Len Sasso*

## The Most Important Tools

The letter "DIY Mastering" that appeared in the April 2007 issue of EM caught my eye, as I am currently using Tannoy speakers and Steinberg WaveLab 5 (with no Mackie board) to master Digidesign Pro Tools sessions from my home. Yes, I said *master*. Without getting into that debate, I must stress that gear is only one ingredient in the production pie.

In my career of over 15 years as a professional engineer, I've worked on projects of all kinds and in many different environments. With each project comes a new tool set to work with. Regardless of the project budget or my personal preferences for specific boutique devices, I must find a way to deliver what the client expects. While I agree with Tony on many levels, I can't help but get the impression that he and many others are overlooking the importance of good ears and solid experience.

I've put in my time at FOH (front of house), taught audio production, and designed many systems and studios. But as a new dad, my golden audio toys were sold to pay for diapers and all that other fun stuff. I do miss them at times, but the most important ones are still in my mind and attached to the sides of my head.

Justin Mai
Boise, Idaho

## Evolution in EM?

The last place I expected to find a reference to human evolution was in EM. But there it was, a big quote on p. 70 of the April 2007 issue (see "Square One: How Your DAW Does Math"): "Had humans developed with two fingers instead of ten, we might well consider binary numbers normal."

"Had humans developed"? I respect the views of others and expect a magazine about recording matters to be sensitive to this as well. No doubt, many of your readers don't think we "developed" ten fingers, but were instead created that way. It would be easy to avoid this by editing the sentence to say, "If humans had two fingers instead of ten, we might well consider binary numbers normal." Such a change wouldn't have altered the effectiveness of a great article. My point is this: there are places where the insertion of one's views is appropriate, and EM is not one of them.

Aaron Sain
Nashville, Tennessee EM

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EMspotlight

## Mixing with Bob Clearmountain

Recording engineer, producer, and mix engineer Bob Clearmountain has a discography that reads like a rock 'n' roll encyclopedia: Bryan Adams, David Bowie, Paul McCartney, Robbie Robertson, and the Rolling Stones, to name a few. In this interview from the EM archives, Clearmountain takes time from his busy schedule to share insights on his mixing techniques, his personal studio, and his illustrious career. By Jeff Casey. emusician.com/em_spotlight

## GET ON THE BUS!

The Bus is the exciting new blog by the EM editors that includes tips and tricks on a variety of recording topics, covers the latest technologies and industry trends, and offers a behind-the-scenes look at trade shows, product demos, and the creation of *Electronic Musician* magazine.
blog.emusician.com/the_bus

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## EM WEB CLIPS

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## SPECIFICATIONS

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By Geary Yelton

## Zoom H2

Demonstrating that pocket digital recorders are getting less expensive and more versatile, Zoom (www.zoomfx.com) has announced the H2 ($334.99), a palm-size 24-bit recorder with three built-in mic capsules. Mics positioned left, right, and center allow directional, omni, or mid-side recording from the H2's front or rear. The H2 gives you a choice of recording angles: 90 degrees, for recording individual voices or instruments; 120 degrees, for recording ensembles on a stage; or 360 degrees, for recording four channels you can convert to 5.1 surround. The maximum sampling rate is 96 kHz for stereo or 48 kHz for 4-channel recording.

The H2 has a built-in USB 2.0 audio interface that lets you use it as a USB microphone and transfer audio to your Mac or PC. It features 1-button recording and has a graphic display, transport controls, and recording-level buttons on the front, with a gain switch and output-level controls on the side. The H2 supports time-stamping and track-marking functions when it's recording in Broadcast WAV Format (BWF), and it also records variable-bit-rate MP3 files at up to 320 Kbps. The H2 records on Secure Digital cards and includes a 512 MB card, a tripod stand, a mic-clip adapter, a USB cable, and an AC adapter.

## Tascam FireOne

The newest desktop wonder from Tascam (www.tascam.com) is the FireOne ($399), a FireWire audio/MIDI interface with control surface functionality. The compact device has two XLR/TRS combo inputs with 48V phantom power and –20 dB pads, a high-impedance ¼-inch input for guitar and bass, MIDI In and Out, two unbalanced ¼-inch outputs, and two ¼-inch stereo headphone jacks with independent level knobs. The FireOne handles 24-bit audio at rates up to 96 kHz, and Tascam promises 192 kHz support in a future software update. Power comes from the FireWire bus or the included 12V adapter.

The FireOne was designed for hands-on control of digital audio sequencers and other software for the Mac and Windows. The weighted jog/shuttle wheel lets you quickly and intuitively scroll through sequences and waveform displays, and the wheel's backlight pulses in tempo with the music. Eight programmable function keys supplement five transport buttons, and a Shift button doubles every function, including wheel and transport-button operations. Two 12-segment LED meters keep track of input and output levels. A ¼-inch footswitch input lets you control tasks such as punch-in and -out and start and stop. The FireOne comes with a copy of Ableton's Live Lite 6 Tascam Edition.

## Arturia Jupiter-8V

The Roland Jupiter-8 was one of the most desirable analog synthesizers of the early 1980s. It was an 8-voice instrument, and each voice had two VCOs with cross-modulation, a resonant lowpass VCF, a nonresonant highpass filter, a 4-waveform LFO, and two ADSR generators. The latest vintage reproduction from French developer Arturia (www.arturia.com) is Jupiter-8V (Mac/Win, $249), a 32-voice polyphonic software emulation of the pre-MIDI classic. Jupiter-8V puts the original's timbral palette on your desktop and adds features such as a selection of multi-effects and a step sequencer you can use as a modulation source.


Jupiter-8V's front-panel layout is practically identical to the original's, but you can open an additional graphic display area for selecting presets and editing the step sequencer, effects, and modulation parameters. A feature called Galaxy lets you combine LFO types into a unique mod source you can control in real time. A collection of more than 400 presets is included. Like most Arturia instruments, Jupiter-8V runs standalone and as a plug-in for AU, RTAS, and VST hosts.

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– ***Sound On Sound*** **Magazine**

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Garritan Gofriller Solo Cello

WHAT'S NEW

## Ueberschall Liquid Trumpet


The seventh addition to the Liquid Instruments series from Ueberschall (distributed by Big Fish Audio, www.bigfishaudio.com) is Liquid Trumpet (Mac/Win, $199.95). It furnishes 1.5 GB of 24-bit licks and phrases—80 loop-based construction kits—in a variety of musical styles, including jazz, funk, Afro Cuban, and bebop. Each construction kit contains six loops with a main theme, variation, solo, and ending. Parts were played on muted and unmuted trumpet and flügelhorn. Liquid Trumpet operates standalone or as an AU, VST, or RTAS plug-in.

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## Rev Up

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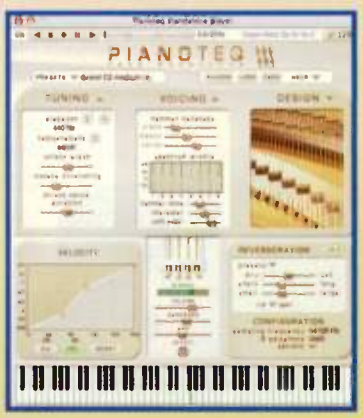
The latest addition to the suite is Mono/Poly, a software reproduction of the analog synth launched in 1981. The original Mono/Poly had four VCOs you could stack together in unison or cycle through in polyphonic mode, each with independent pitch, level, and waveform. Its architecture also featured a noise generator, a multimode filter, two ADSR generators, two LFOs, Synchro- and cross-modulation, chord memory, and a built-in arpeggiator. The software version adds 128-voice polyphony, up to 16-voice unison, 256 new patches, and an 8-way modulation matrix with 159 sources and 35 destinations. Korg Legacy Collection - Analog Edition 2007's instruments and effects run standalone and support AU, RTAS, and VST plug-in formats.

### MODARTT PIANOTEQ 2

A new and improved version of the virtual piano from French software developer Modartt (www.pianoteq.com) is now available. Like the original, Pianoteq 2 (Mac/Win, $337) uses physical modeling rather than sampling to capture the intricate sound and nuance of a fine grand piano. The result is a virtual instrument that requires only 15 MB of disk space and loads quickly, yet sounds surprisingly lifelike.

In addition to operating as an AU or VST plug-in, Pianoteq 2 runs standalone. A new soundboard model engenders even greater realism, as demonstrated by the Grand C2 collection of presets. You can vary the lid position from open to half-open to closed, and use the Pan slider to alter the stereo image's width. If you like to experiment with microtuning, Pianoteq 2 furnishes 30 presets ranging from Wendy Carlos's Alpha scale to pre–Baroque Zarlino. You can use predefined MIDI Control Changes for smooth, hands-on control of 19 parameters such as piano size, hammer hardness, and sympathetic resonance. Pianoteq now responds to Release Velocity, and new reverberation presets let you alter the perspective from player to audience. EM



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# Garritan Gofriller Solo Cello


Garritan (www.garritan.com) has released Gofriller Solo Cello (Mac/Win, $199), the second virtual instrument from the same team responsible for Stradivari Violin 2.0. Like that software, Gofriller Solo Cello is based on Native Instruments Kontakt Player 2 and incorporates expressive techniques developed by Giorgio Tommasini. It allows you to control real-time articulation using various MIDI Control Changes; for example, you can change the bow direction and dynamics, play trills and glissandos, and control the onset, rate, and intensity of vibrato, much as if you were playing an acoustic instrument.

Matteo Gofriller was an early-18th-century Italian stringed-instrument maker especially renowned for the rich and beautiful tone of his cellos. Garritan sampled one of those fine instruments and applied proprietary harmonic-alignment and sonic-morphing techniques to bring the samples to life by allowing samples to crossfade across dynamic layers. In addition to standalone operation, Gofriller Solo Cello runs as a plug-in for AU, DXi, RTAS, and VST hosts.

# Download of the Month

## BROKEN LIGHT PLUG-INS (WIN)

PhoneMon 1.0 is one of several freeware plug-ins from Bermudan software designer Ralph Gonzalez (www.geocities.com/rgonzale1/audio/vst). PhoneMon is inserted in the master channel of your audio sequencer to emulate four common listening environments through your studio speakers or headphones: Earbud (light bass and treble), Boombox (mid-bass peak), 2" Speaker (light bass and treble cut), and Boom & Tizz (peaky bass and treble). PhoneMon also sports a simple reverb and low- and high-frequency trim controls for headphone compensation. You can get similar results from an EQ plug-in, but PhoneMon is fast and does a credible job.

Badamp 2.1.2 is a guitar amp-and-cabinet simulator. It lacks the versatility and polish of high-end simulators like Native Instruments Guitar Rig 2 and Line 6 Amp Farm, but all the right stuff is lurking under the hood. Multimod 1.2 is a tremolo and delay-based effect. You get sine-, triangle-, and square-wave tremolo (amplitude modulation) shapes with depth and frequency controls. Two feedback delay lines with modulators supply a variety of flange, phase, and resonator effects.


NewWave and Static round out Broken Light's effects offerings. Static is a threshold-based noise generator. Signals above the threshold level cause noise bursts proportional to the triggering signal's level. Lo Cut, Hi Cut, and Sparse knobs shape the noise; the Sparse knob affects granularity, producing anything from static to white noise. NewWave is another threshold effect. It replaces incoming audio with a square, sine, or triangle wave if it's above the threshold level and within a passband set by the Lo Cut and Hi Cut controls. You can transpose the output's pitch and mix in a second oscillator with its own pitch offset. The pitch tracking is pretty touchy, but the results are interesting if you're not after precision.

Broken Light offers two basic synths. Noiseonator is a noise generator with a MIDI-controlled bandpass filter. MIDI Notes set the filter's center frequency, and an ADSR envelope shapes the noise further. Drumlet is a pulse-wave and noise drum synth with built-in feedback delay. Although it produces only one sound at a time, you can restrict its MIDI Note range and set triggering to Note On or Note Off messages. If your host permits it, you can run several instances of Drumlet in parallel to build a viable drum machine.

All the Broken Light plug-ins were created in Synthedit and have a home-brew quality. Documentation is sparse, usually limited to short About notes accessed from the control panel. No guarantees are given or implied, but the price is nice.

*—Len Sasso*

# Terratec Producer Axon AX 50 USB


Many MIDI guitarists consider the Axon AX 100, from Terratec Producer (distributed by Synthax, www.synthax.com), to be the most accurate guitar-to-MIDI controller available, but its wealth of advanced features may be more than you need. Instead of a built-in synth, sequencer, and arpeggiator, the Axon AX 50 USB ($549) has something else guitarists have requested: USB connectivity. It also has a 3-character LED display, a built-in tuner, an output for your guitar's magnetic pickup signal, and MIDI In, Out, and Thru ports. The Axon AX 50 USB includes Kontakt Player, which comes with 128 sample-based instrument presets. All you need is a guitar or bass with a hexaphonic pickup and your computer or MIDI instrument.

The unit's 32-bit RISC processor translates the vibrations of guitar strings into data that captures the pitch and dynamics of a guitarist's performance. In addition to being able to play an external MIDI instrument, the Axon AX 50 USB can play several software instruments at the same time, thanks to four virtual MIDI ports made possible by USB 1.1. For real-time multitimbral performances, the controller divides your fretboard into as many as 12 zones. The Axon AX 50 USB stores 128 user-programmable presets and comes with a software editor for the Mac and Windows.

# BIAS Master Perfection Suite


When Berkley Integrated Audio Software (www.bias-inc.com) introduced Peak Pro XT 5, it included a plug-in bundle that extended its capabilities. Now BIAS has begun shipping an AU- and VST-compatible edition of the Master Perfection Suite (Mac, $599). BIAS promises that RTAS and Windows VST compatibility will follow later this year. Peak Pro XT 5 owners can download the new Master Perfection Suite for free.

GateEx is a noise gate and downward expander. As a gate, it reduces an audio signal's amplitude when it falls below a threshold you define, and as an expander, it increases a signal's apparent dynamic range. PitchCraft is a real-time pitch-correction and -transposition plug-in that incorporates formant shifting to retain or modify vocal characteristics. The spectral-matching plug-in Repli-Q can analyze and compare two audio recordings and apply the harmonic characteristics of one to the other. It can also compensate for spectral problems and restore balance to a mix. Reveal is a suite of tools for analyzing frequency, level, and phase. It includes an oscilloscope, a spectrogram, a Lissajous phase scope, peak and RMS level meters, and more. SuperFreq is a collection of paragraphic EQ plug-ins with 4-, 6-, 8-, and 10-band variations. For multiband compression, limiting, and upward expansion, Sqweez-3 and Sqweez-5 furnish three and five bands, respectively.

# KRK Systems VXT Series

Speaker manufacturer KRK Systems (www.krksys.com) has introduced three new models in its line of biamplified close-field studio monitors: the VXT4 ($399), VXT6 ($599), and VXT8 ($799). Taking its design cues from KRK's flagship Exposé E8B, each model features a proprietary woofer and tweeter, a curved faceplate for superior imaging, and a low-resonance cabinet with slotted ports to reduce port turbulence. The cabinets are constructed of sturdy ABS foam, which offers excellent damping characteristics and makes them impact resistant. Dome tweeters are made of silk for a fast transient response and good imaging, and woofers have a woven Kevlar cone for low distortion and an extended low end.

The VXT4 has a 4-inch woofer and weighs 14 pounds; it produces 30W of continuous power into 4Ω at the low end and 15W into 8Ω at the high end. The VXT6 has a 6-inch woofer and weighs 27 pounds; it's rated at 60W on the bottom and 30W on top. With an 8-inch woofer, the VXT8 weighs 41 pounds and maxes out at 120W and 60W. All three monitors have 1-inch tweeters and XLR/TRS combo input jacks.

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## KORG LEGACY COLLECTION - ANALOG EDITION 2007

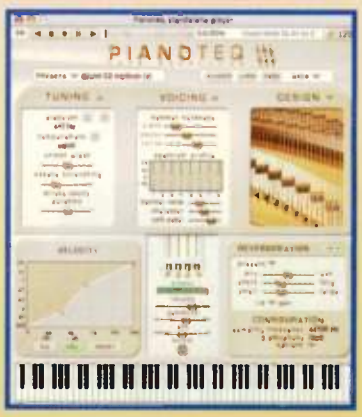
Korg (www.korg.com) is shipping a new version of the Legacy Collection, a software bundle that emulates classic Korg synthesizers and effects. The Legacy Collection - Analog Edition 2007 (Mac/Win, $299) reconfigures the bundle to focus exclusively on analog instruments. It includes the MS-20 and Polysix virtual synths, MS-20FX and MDE-X effects processors, and Legacy Cell, which combines the two synths into a single instrument. Wavestation, however, is now part of the Legacy Collection - Digital Edition.


The latest addition to the suite is Mono/Poly, a software reproduction of the analog synth launched in 1981. The original Mono/Poly had four VCOs you could stack together in unison or cycle through in polyphonic mode, each with independent pitch, level, and waveform. Its architecture also featured a noise generator, a multimode filter, two ADSR generators, two LFOs, Synchro- and cross-modulation, chord memory, and a built-in arpeggiator. The software version adds 128-voice polyphony, up to 16-voice unison, 256 new patches, and an 8-way modulation matrix with 159 sources and 35 destinations. Korg Legacy Collection - Analog Edition 2007's instruments and effects run standalone and support AU, RTAS, and VST plug-in formats.

## MODARTT PIANOTEQ 2

A new and improved version of the virtual piano from French software developer Modartt (www.pianoteq.com) is now available. Like the original, Pianoteq 2 (Mac/Win, $337) uses physical modeling rather than sampling to capture the intricate sound and nuance of a fine grand piano. The result is a virtual instrument that requires only 15 MB of disk space and loads quickly, yet sounds surprisingly lifelike.

In addition to operating as an AU or VST plug-in, Pianoteq 2 runs standalone. A new soundboard model engenders even greater realism, as demonstrated by the Grand C2 collection of presets. You can vary the lid position from open to half-open to closed, and use the Pan slider to alter the stereo image's width. If you like to experiment with microtuning, Pianoteq 2 furnishes 30 presets ranging from Wendy Carlos's Alpha scale to pre-Baroque Zarlino.




You can use predefined MIDI Control Changes for smooth, hands-on control of 19 parameters such as piano size, hammer hardness, and sympathetic resonance. Pianoteq now responds to Release Velocity, and new reverberation presets let you alter the perspective from player to audience. EM

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iO|14 FireWire

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# Thanks for the Memories

By Scott Wilkinson

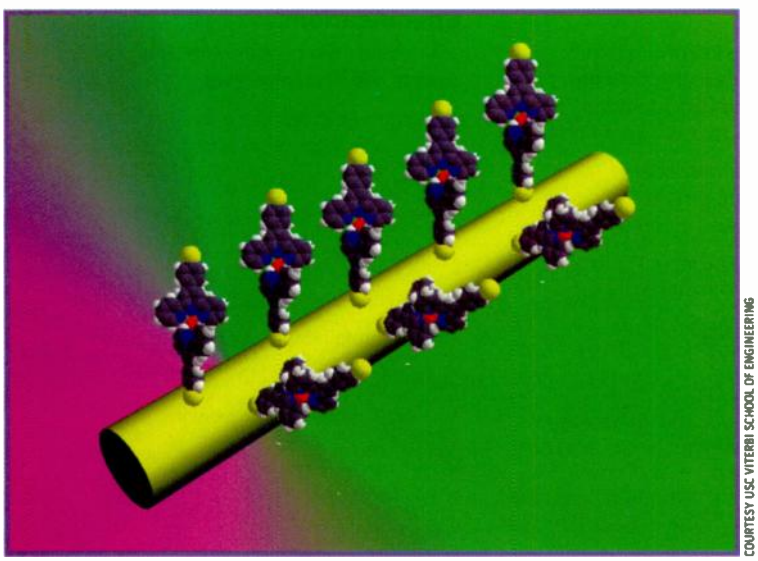
## Molecular memory comes closer to reality.

As electronic musicians, we have insatiable appetites for digital storage; no matter how much we have, we always need more. Hard disks and solid-state storage (flash memory) keep increasing in capacity while decreasing in cost, but that trend can't continue forever.

Or can it? If research that is being conducted at the University of Southern California's Viterbi School of Engineering (http://viterbi.usc.edu) and the NASA Ames Research Center in Palo Alto, California, comes to fruition, solid-state storage capacity could expand dramatically beyond what is possible with conventional flash memory.

USC associate professor Chongwu Zhou leads a team that has demonstrated the potential of nanotubes as the basis for molecular memory devices. The nanotubes created by Zhou's team are made of indium oxide and are about 10 nanometers (nm) in diameter and 2,000 nm long. To give you a sense of scale, an average bacterium is about 1,000 nm long, and the smallest known virus is about 20 nm long. One nanometer is about the width of ten hydrogen atoms.

In a process called laser ablation, the nanotubes self-assemble in an oxygen-rich environment after indium is vaporized. They are then placed on a thin layer of quartz and submerged in a special material that coats them. Molecules of the material (in this case, iron terpyridine) attach themselves to the nanotubes (see **Fig. 1**), forming a field-effect transistor that can act as a memory cell.



COURTESY USC VITERBI SCHOOL OF ENGINEERING

**FIG. 1: Molecules of iron terpyridine attach themselves to a nanotube of indium oxide, forming a memory cell capable of storing three bits of data in far less space than silicon memory cells.**

Perhaps the most important property of the coated nanotubes is that they can assume one of eight charge states, allowing each one to store three bits of digital information ($2^3 = 8$). The semiconducting nanotube is exquisitely sensitive to these states, changing its conductivity accordingly. As a result, the data can be read by sampling the resistance of the nanotube.

Not only can each memory cell hold three bits of data, but the cells are also much smaller than conventional silicon cells, leading to an initial projected data density of 40 GB (a little more than a DVD's worth of data) per square centimeter. Compare that with 1.8 GB/cm² in current silicon memory and 10 to 20 GB/cm² in today's hard-disk drives. And that's only the beginning, according to Zhou. "If we scale down the nanowire length by a factor of 10 to 200 nanometers," he says, "the density can approach 400 GB per square centimeter."

Another breakthrough in Zhou's research is the stability of the memory cells. Whereas most 1-bit-per-cell molecular memory prototypes retain their data for no more than a few hours, Zhou's nanotube cells can hold on to their data for up to 600 hours. Of course, that's not nearly long enough to be commercially useful, but it's certainly a step in the right direction.

Other potential advantages of these nanotubes include low manufacturing cost (the devices assemble themselves) and much lower power consumption than silicon memory. Still, Zhou believes that practical applications are five to ten years away. "Substantial development work is needed to push the performance even further and to develop a fabrication process amenable for mass production," he says.

The benefit to electronic musicians is obvious. Flash memory cards with 20 to 200 times the storage capacity of today's cards could easily hold many long tracks of high-resolution audio, lots of high-definition video, and endless MIDI data in a portable device with no moving parts. What more could we possibly want? (Ask me again in 20 years!) EM

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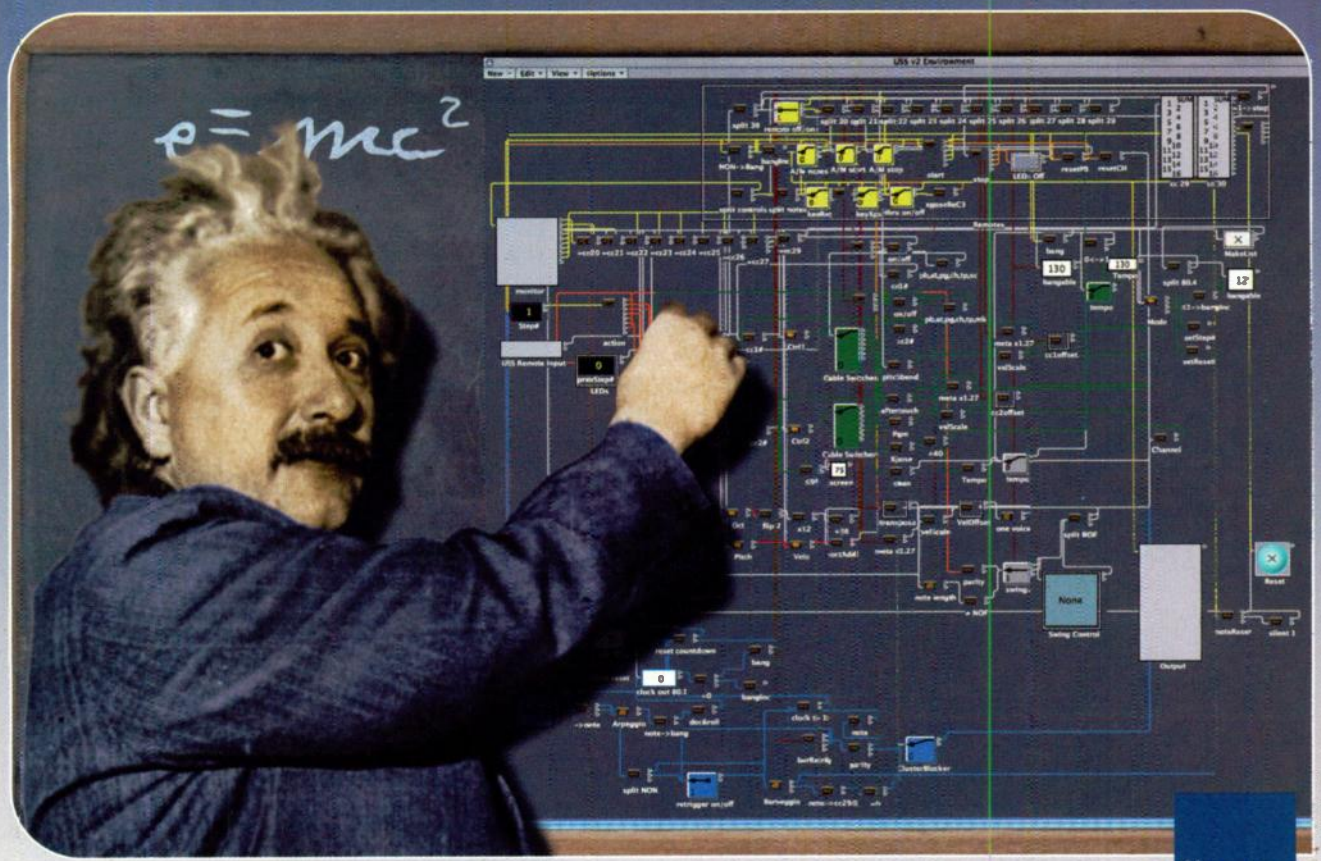


LAURA WILLIAMS

Master CLASS

# Think Different

By Len Sasso

## The logic behind Apple Logic Pro.

Apple Logic Pro 7.2 has features that are the envy of its competitors, but the implementation of those features is not always completely intuitive. In some cases, Logic reaches beyond the standard audio-sequencing paradigm. In others, a new process doesn't fit neatly within Logic's structure, and its implementation may be somewhat ad hoc. In this article I will attempt to demystify some of the more common stumbling blocks to mastering Logic.

Like most sequencers, Logic has settings that apply to all songs (preferences), settings that change from song to song (song settings), and user-assignable keyboard shortcuts (key commands). It's tempting to think of preferences and song settings as set-and-forget items, but you can use them interactively, and I'll mention several useful song settings as I go along.

I won't cover specific key commands except the default ones for opening the Key Commands window (Option + K) and the Preferences window (Option + P). It's a good idea to assign one to open the Record Song Settings window because you may want to change the settings on that page frequently, and you can access all other song settings from that window. You'll find key commands for just about every Logic menu item as well as for many functions not accessible in any other way. Customizing Logic's key commands is one of the best ways to improve your work flow.

### The E Word

Other digital audio sequencers don't have anything resembling Logic's Environment, or better said, they keep it out of reach. You don't need to know a great deal about the

Environment to use Logic, but a basic grasp is essential.

In most sequencers, a new track comes with all the attributes you need in order to use it. In Logic, a track is simply a lane in your arrangement; to make the track useful, you need to assign an object from the Environment to it (which happens automatically when you create the track). That gives you a great deal of freedom at the price of a little extra management. For example, if you want to change sounds on a virtual instrument track or change effects on an audio-playback track, you can have several Environment objects set up with alternative plug-ins and then simply switch the track assignments. That's faster and easier than loading alternative channel-strip presets, though you can take that approach in Logic as you can in other sequencers.

If you use MIDI gear outside of your computer, including MIDI instruments running on another computer, you need to know a little about the Environment's MIDI instrument objects: Instrument, Mapped Instrument, and Multi-Instrument. If you slave other applications to Logic using Propellerhead ReWire protocol, you need to know about the Internal ReWire object (MIDI out) and the Audio ReWire object (audio in). There are a variety of Audio object types, but the ones you primarily need to know about are Output, Track, Instrument, and Aux.

## E for Audio

Output objects feed your audio hardware outputs and are also used for bouncing. Track objects are used to record and play back audio on arrangement tracks. Instrument objects host virtual instrument plug-ins; their arrangement tracks record and play back MIDI. Both Track and Instrument objects host effects plug-ins and have bus sends.

Aux objects take their input from send buses, Logic's external inputs, or multioutput virtual instrument plug-ins. They are primarily used to host send effects. Two other objects, Bus and Input, are not required but are useful for, respectively, isolating a single channel on a stereo bus and destructive effects processing of incoming audio.


FIG. 1: Seven arrangement tracks are assigned to Logic's Ultrabeat drum synth. The kick and snare are routed to separate outputs handled by the two Aux tracks at the bottom.


FIG. 2: In Contents Catch mode, the Matrix Edit window reveals the contents of the MIDI region currently being played.

Bear in mind that you can assign the same Environment object to several tracks—a handy feature as well as a common source of confusion in Logic. Most of the confusion is due to the Mute and Solo buttons that appear on both the object and the track. The buttons on the object mute and solo all tracks assigned to the object, whereas the buttons on the track mute and solo only that individual track.

Assigning the same Audio object to several tracks is handy for managing different takes, maintaining separate tracks for different sounds on multitimbral instruments such as drum synths, using Logic's folder hierarchy, and managing automation, among other things (see Fig. 1). Although audio regions can occupy the same time position on different tracks assigned to the same Track Audio object, only one region can play at a time, and that's the one most recently encountered by the Song Position Locator. To crossfade, put regions on the same track; to play simultaneously, put them on tracks assigned to different Track objects.

## E(xceptions)

Arrangement tracks are assigned to Environment objects by clicking on the track name and selecting the Environment object from the pull-down tracklist. The bottom two choices, No Output and Folder, are not Environment objects. No Output tracks are handy for holding MIDI and audio that you want disabled but do not want to delete. They're also handy for recording

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"feedback" Environment processes, but that's another story.

Folders are your access to Logic's hierarchical approach to sequencing. Many sequencers allow you to group tracks and then hide the individual group elements in order to unclutter your arrangement. Logic brings a new dimension to that process by nesting whole arrangements within arrangements. Folders can hold any number of tracks along with their contents, and you can locate them anywhere on Logic's timeline. They're great for holding submixes, tracks for multitimbral instruments, and alternative arrangements. When you pack a folder, it is automatically placed on the lowest Folder track in the arrangement, and if there are no Folder tracks, a new one is created. Folder tracks are a convenient place to put folders, but a folder will actually play back on any track except one assigned to No Output.

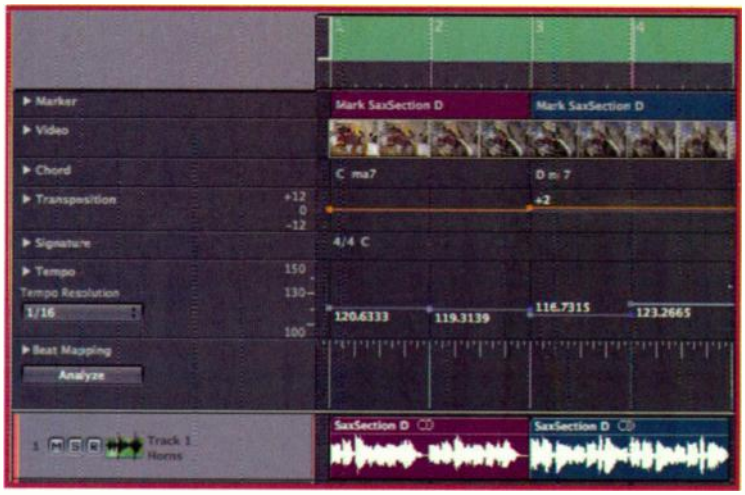


FIG. 4: Global Tracks at the top of the Arrange window manage markers, video thumbnails, chord names, Apple Loop and MIDI file transpositions, tempo changes, and beat mapping.

Logic's Audio window, Sample Editor, and Project Manager all use the same Track Audio object for auditioning audio files. You can select which object is used, but the selection is global and reverts to the Track 1 Audio object whenever Logic is relaunched. So it's easiest to let the Track 1 Audio object be devoted to auditioning and to avoid using it in arrangements. Otherwise, its effects plug-ins and automation will interfere with auditioning. In a bit of particularly obscure reasoning, the highest-numbered Track Audio object rather than Track 1 is used for auditioning in the Apple Loops browser, so it's a good idea to also reserve that object for auditioning. I keep both objects assigned to arrangement tracks in order to have quick access to auditioning levels in the Arrange and Mixer windows.

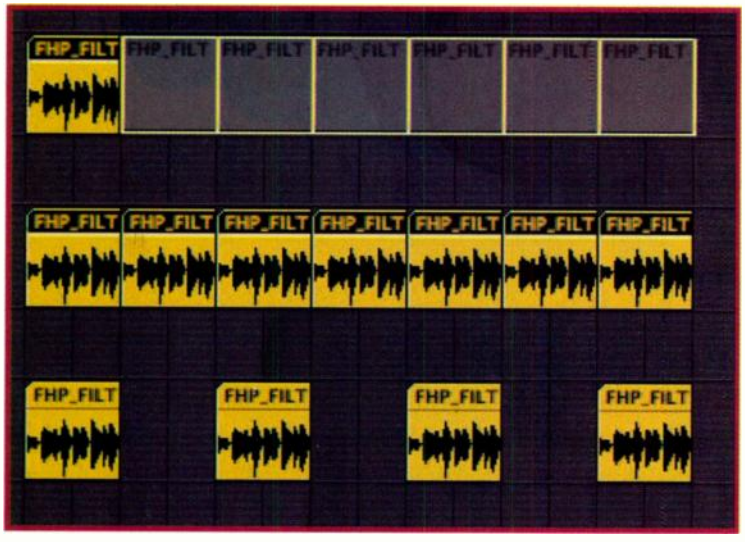


FIG. 3: To create alternating loops of a region, turn looping on, convert the loops to real copies, and delete the unwanted copies.

## Logic Does Windows

In Logic you can open multiple instances of the same window in the same Screenset. For instance, you could have several Arrange windows showing the contents of different folders, or you could have several Matrix Edit windows showing the contents of different MIDI regions.

The Link button (with the chain icon) at the top left of each window determines how the window behaves relative to the currently active (topped) open window. It has three modes: Off (no linking), Link (pink—shows objects selected in the active window), and Contents Link (gold—shows the contents of objects selected in the active window). As a simple example, if you have two Arrange windows open and both are in Contents Link mode, then whenever you select a folder in either window, the other window will reveal the contents of that folder. Linking and contents linking can seem a little counterintuitive at first, but they're a huge time-saver when using multiple windows of any kind.

The Catch button (with the running-man icon) invokes the follow-song option familiar in most sequencers. When it's used in conjunction with Contents Link mode (which is called Contents Catch mode), the contents of any region on the selected track are revealed in the contents-linked window whenever the song position passes over it (see Fig. 2). That's another big time-saver.

Logic's Arrange window and three MIDI Editor windows—Event List, Matrix Edit, and Score—are hierarchical. Clicking on the Up Level button in the upper-left corner of these windows moves you up one level in the hierarchy. For the Arrange window, that moves you up through nested folders. For the Event List window,


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it moves you both up through nested folders and up from showing MIDI messages to showing the regions that contain them. For the Matrix Edit and Score windows, it moves you from showing the contents of a single MIDI region to showing the contents of all MIDI regions. When viewing multiple regions in the Matrix Edit window, turn Region Colors on (in the View menu) and ensure that the regions have different colors. Then you can see which notes belong to which regions.

You can open any window as a floating window by holding the Option key when selecting it from Logic's Windows menu. Most windows have a two-dimensional Scroller to the left of the horizontal scrollbar, which you use to scroll either horizontally or vertically by dragging in the corresponding dimension. You can set up key commands to store and recall three zoom settings as well as to navigate backward and forward through recent zoom settings. Finally, you can click-and-drag an object to a new location in a window without topping that window. (Only a short click tops a window.)

## Regional Differences

Unlike audio and MIDI, global data such as tempo, time signature, key signature, markers, and video is not contained in regions. Logic's primary form of automation is also not held in regions, although you can use regions for automation.

Logic gives you some useful options when doing cycled overdub recording. You can choose to have the previous pass muted when starting a new pass, and you can have Logic automatically create a new track for each new pass. The new track will use the same Environment object, but when you're done you won't have a bunch of recordings stacked on top of each other, and you can use the track Solo buttons to audition the various passes.

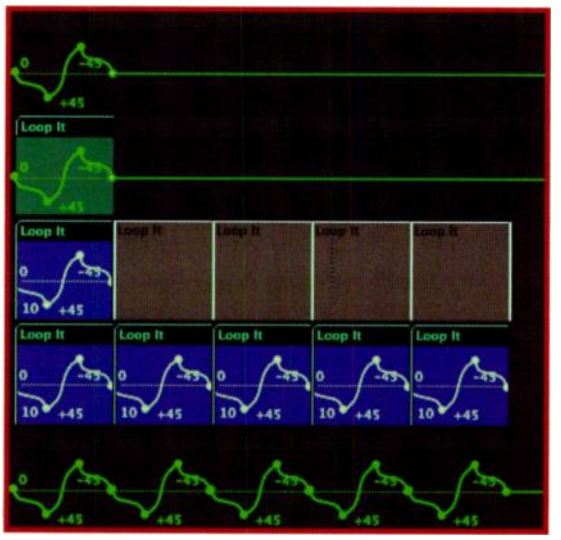


FIG. 5: Five steps to looping automation: create one loop of automation, place an empty MIDI region under it and transfer the automation to the MIDI region, loop the MIDI region, turn the loops to real copies, and transfer the automation back to the track.

For MIDI recording, you have a couple of additional options. You can choose to have each new pass merged with the previous passes, leaving you with a single MIDI file containing all the data. If you're recording from several different MIDI controllers on different MIDI channels, you can have Logic automatically split out the separate MIDI channels when you stop recording. All those choices are found in the Record Song Settings window, which is why it's handy to have a key command to open it.

Here are several things to know when you do wind up with overlapping regions in the Arrange window, whether as a result of recording or of dragging regions around. If you click on some location outside of the overlapping regions, they will be redrawn with regions later in time shown on top. You can simultaneously create new tracks for all overlapped regions from the Arrange window's Track menu. You can choose Remove Overlaps from the Region menu to have Logic automatically truncate all selected regions to remove any overlaps. (Drag-select from right to left to select all overlapped regions on a track.)

Two pull-down menus at the top right of the Arrange window affect dragging regions. The Snap menu determines how dragging is quantized, and the Smart option at the top of the menu invokes zoom-dependent snapping (zoom in for more-precise dragging). Keep in mind that what is quantized is the size of the drag steps, not the absolute position. For example, if the snap is set to Bar, dragged regions will jump one bar at a time, not snap to the next bar line. To override the snap settings, hold the Control key (quantize to Logic's display format) or hold the Control and Shift keys (no quantize—move in ticks).

When you drag regions over each other, the behavior is set by the Drag menu: Overlap is the normal behavior; No Overlap truncates the region that is dragged over; X-Fade introduces an automatic crossfade the length of the overlap; and the Shuffle options shuffle the regions' positions while keeping them butted up (try it, you'll like it).

## Loops, Aliases, and Real Copies

You can, of course, make copies of regions by the usual methods, but for MIDI regions you also have the option of creating aliases. As in the Mac Finder, they are references to the original region. (There are no aliases for audio because any audio region is already a reference to the original audio clip.) To create an alias by dragging, hold down the Option and Shift keys. The advantage to an alias is that it reflects any changes you make to the original. The disadvantage is also that it reflects any changes you make to the original, but you can convert any alias to a real copy when you want to edit it individually (MIDI menu).

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You can loop any audio or MIDI region by clicking-and-dragging on its upper-right corner. Loops are always contiguous, but you can convert them to aliases or real copies when you want to separate them (Region/Parameters menu). That's especially handy, for instance, when you want aliases or copies at regular intervals such as every other instance of a loop (see Fig. 3).

## Freeze, Bounce, or Export?

Logic offers three ways to render audio and virtual instrument tracks to audio files: freezing, bouncing, and exporting. Freezing is the fastest but the least flexible. It applies to individual tracks, and it greatly reduces the CPU load by eliminating effects and virtual instrument processing. To freeze a track, click on the track's snowflake-icon Freeze button. (If you don't see Freeze buttons, use the View menu to make them visible.) Freeze files are 32 bit and can't be imported into other Logic songs.

Bouncing applies to all audio that is directed to a specific output, and it is accomplished by clicking on the Bounce button on any Output Audio object. Bouncing supports a variety of file formats (AIFF, SDII, WAV, MP3, AAC, and so on) and supports mono, stereo, and surround. Real-time bouncing renders external audio sources routed through Input Audio objects, whereas the faster, offline bouncing renders only internal audio.

Exporting individual tracks and regions as audio files is the most flexible option, and it's the quickest way to create stems. But be aware that when exporting or bouncing tracks with plug-ins that introduce latency, such as look-ahead limiters, the beginning of the track will have a bit of silence corresponding to the latency. After that, the tracks will be in sync, so you can avoid the problem by having a short, empty lead-in. (Special thanks to Logic Certified Trainer Jay Asher for this tip.) You can export selected regions, the selected track, or all tracks at once. You choose a PCM file format (AIFF, WAV, or SDII) and a bit depth (8, 16, or 24) in the Export dialog box. Volume and pan automation are not rendered when exporting tracks and regions; for that, you need to bounce.

## Global Concerns

All of Logic's time-based windows—Arrange, Matrix Edit, Score, and Hyper Edit—have an optional display at the top for seven Global Tracks: Marker, Chord, Transposition, Signature (Key and Time), Tempo, Beat Mapping, and Video. The Chord, Transposition, and Key Signature tracks work together to determine the automatic transposition

of Apple Loops and MIDI regions, although you can turn off MIDI region transposition in the track's instrument parameters.

You can use the Beat Mapping track in conjunction with the Video track or selected audio or MIDI regions to align hitpoints to meter grid lines. Logic accomplishes that by automatically adding tempo changes to the Tempo track (see Fig. 4). You can, of course, add tempo changes manually, and two special windows—Tempo List and Tempo Operations—allow numerical and graphical tempo editing. Although there is only one Tempo track, you can maintain ten alternative tempo lists and switch between them freely.

The Marker track is for both taking notes and marking song positions. You can place markers manually or have Logic do it automatically by marking selected regions or scenes detected in the movie on the Video track. You can assign key commands for jumping between markers. Use the Marker List and Text windows to manage and edit markers.

## Automation

Logic's track automation is powerful yet extremely easy to use. Each track can have as many automation lanes as needed. You can create automation graphically with the mouse, or you can use a control surface to record automation on the fly. You can automate any track or plug-in parameter. Track automation is not contained within regions, but you can set a preference to have it automatically moved with overlying regions.

You can freely move track automation to regions and vice versa. If you move automation to an audio region, you need to open it in the Event List editor to view the automation. You can view and edit track automation directly in its own list-style editor, but you need to assign a key command to open that editor.

Two kinds of events are used for automation in Logic: MIDI Control Change messages and Logic's special Fader messages. MIDI messages are used for standard parameters such as volume, pan, sustain pedal, and so on. Fader messages are used for plug-in-specific parameters, although if you have a MIDI controller mapped to a plug-in parameter, you can use that as well. You cannot use MIDI CC 7 (Volume) and CC 10 (Pan) for plug-ins, because Logic always intercepts those messages to use for channel strip volume and pan. You can view and edit Fader messages in both the Event List and Hyper Edit windows.

Owing to Logic's excellent graphical tools for editing automation, track automation is usually preferable to region automation. But region automation has its uses, and one of them is automation looping. To loop a chunk of track automation, create an empty MIDI region the length of the desired loop and move it under the track automation. (You can move the MIDI region to any automation lane.) Move the track automation to the region and loop the MIDI region as needed. If you want to convert it to looping track automation, turn the loops to real copies and move the automation back to the track (see Fig. 5).

## The Details

I've tried to point to processes and features that differentiate Logic Pro 7.2 from other digital audio sequencers. Needless to say, the devil is in the details, most of which I've glossed over. *Logic Pro 7 Reference Manual* is an excellent source. A quick search of the PDF version of that tome will almost always get you close enough to figure out the rest for yourself. When that's not enough, the various second-source books and the users forums that the Apple Web site links to are great resources. The answer is out there somewhere. EM

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# Powerhouse Samplers

By Geary Yelton

When you think of electronic musical instruments, one of the first that probably comes to mind is the sampler. Even when you play a synthesizer, chances are good that its sound sources were sampled. When you install a software instrument, it's most often a sample player of some kind. Samplers are everywhere. Since dedicated samplers first appeared in the early 1980s, they have changed the way that music is produced almost as much as digital multitrack recording has.

In the 20th century, samplers were defined by their ability to record snippets of sound and assign them to notes on a keyboard. You just plugged in a microphone or line input, armed and triggered sampling, and placed the recording into a keymap. Now that you have so many ways to get sound into your computer, though, a sampler's ability to actually record samples is beside the point. Today, the difference between a sampler and a sample player is that the sampler allows you to assign your own audio files to MIDI notes. That's something you can't do in sample players. Of course, any real sampler lets you do more than map keys. You can usually edit waveforms to varying degrees, even if it's only defining loops and indicating start and end points.

## Who Needs a Sampler?

Given that most recent sample libraries are bundled with sample players, the first question you might ask is whether full-fledged samplers are really necessary for music production. There are many factors to consider, including convenience, cost, flexibility, ease of use, and efficiency of resources. Let's begin with the latter and note that a multitimbral sampler usually puts fewer demands on your CPU, hard drives, and RAM than a dozen other sample-based software instruments. With a sampler, you don't need to learn and navigate a myriad of user interfaces, and you don't need to purchase a software framework you already have every time you add to your sample library.

Samplers allow you to tweak and extend a sound's parameters in ways that mere sample players can't. Because sample players limit your access to parameters, they necessarily limit your creativity, whereas a good sampler stimulates creativity. Want to make bees buzz in harmony? You can do that with a sampler.

Want to play a trumpet two octaves below its normal range? No sweat. Want to transform a purring kitten into a growling gorilla? With a little effort, you can do that, too. When it comes to sound design, you just can't beat samplers.

Samplers are useful for every musical genre—not just electronic music, and not just the music that's most popular. If sample collections aren't available in the latest experimental genre and that's your style, you can create your own or twist available sounds enough to make them your own. Samplers ensure that your timbral palette is yours and yours alone. After all, musical sound is much more than the notes you play. Researching this article has convinced me that the audio-editing power inherent in samplers and the unique capabilities they bestow on their users will increase their influence on the future direction of music.

## Different Strokes

Because so much software is available, I had to narrow the focus of this article so that it wouldn't fill the entire magazine or become diluted. I deliberately excluded sample players such as IK Multimedia SampleTank 2.2, which don't let you map note and Velocity zones. I left out percussion-oriented samplers such as Native Instruments Battery 3, which focus specifically on organizing drum kits. And I did not include loop samplers such as iZotope pHATmatik Pro, which don't provide tools for assembling traditional multisampled instruments. The sample-player category alone would have added dozens of products to the lineup, and I had to draw the line somewhere.

Because so many lesser samplers from small developers are available, I decided to concentrate on the 11 top samplers from well-known companies (see the sidebar "Forward into the Past"). Three run only in Windows, one runs only on the Mac, and the rest are cross-platform. Ableton Sampler, Apple EXS24, Digidesign Structure, and Propellerhead Reason NN-XT work only within their specific hosts. Of the remainder, Cakewalk DS864 is a DXi plug-in and Tascam GigaStudio Orchestra is standalone only; more than half run standalone and also support more than one plug-in format. Prices range from $199 for Ableton Sampler to $599 for

## FORWARD INTO THE PAST

Six years ago, Associate Editor Dennis Miller and I wrote an EM cover story detailing the state of the art by exploring 11 software-based samplers (see "Soft Sampling" in the October 2001 issue, available online at www.emusician.com). Obviously, a lot has changed. Only two of the samplers covered here even existed then. Popular programs such as Bitheadz Unity and Digidesign SampleCell have vanished completely. Hardware samplers have all but fallen off the map. Computers have grown far more powerful, enabling software to do things that hardware never dreamed of, like playing massive multisamples from RAM or streaming 2 GB pianos direct from high-speed 500 GB hard disks. Back then, we were concerned that desktop computers didn't have enough zip to offer reliable polyphony, and that audio file formats were all over the proverbial map. Control surfaces were still uncommon in 2001, giving hardware the edge for tactile control.

Of all our concerns, the one that remains today is obsolescence. When new operating systems replace old ones, or when new processors send old hardware to the scrap heap, there's a risk that software won't run on updated computers unless the software developer updates the sampler too. On more than one occasion, I've avoided retiring an old computer simply because I wasn't willing to stop using perfectly good software that was no longer current (Antares Infinity, Korg OASYS PCI, and Opcode Galaxy come to mind). Still, if you stick with a developer you trust to be around in a few years, and your software is popular enough to have a wide user base, you should feel almost as confident that your software will work in ten years as you are that your hardware sampler won't break and parts will continue to be available.

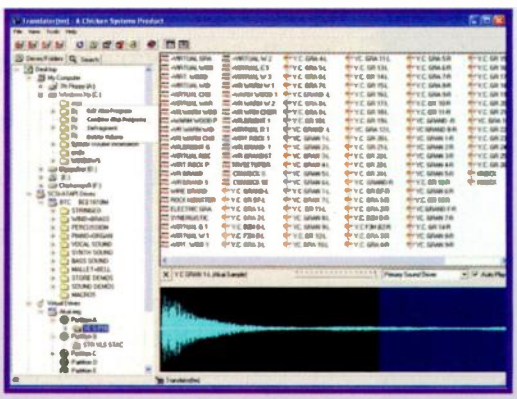

FIG. A: Although some keyboard samplers can import samples and SoundFonts, you'll need Chicken Systems Translator Pro to transfer instruments from software samplers to hardware.

Lest you think that hardware samplers have become obsolete, they still serve a crucial function: playing samples onstage. Many modern synthesizers have the ability to import user samples, but you'll need a file translation utility to export mapped sampler instruments from software (see **Fig. A**).

GigaStudio Orchestra. EXS24 mkII is an integral part of Apple Logic Pro 7, which retails for $999.

### Ableton Sampler 1.0 (Mac/Win)

Sampler ($199) is an optional add-on that first shipped in mid-2006 and runs only in Ableton Live 6. Given the small size of its graphical user interface, Sampler's wealth of user parameters makes it surprisingly powerful. Outstanding features include five envelope generators (EGs), three LFOs, and the ability to modulate sample-playback characteristics and morph between filter types in real time. Because of the way Live integrates its various functions, you can record a sample, drag it into Sampler, and immediately begin processing it into a playable instrument. Ableton relies on serial numbers for copy protection; Sampler requires its own serial number in addition to authorization for Live.

When you open Sampler, it appears in Track view at the bottom of Live's main screen. Sampler has six views you access with tabs on its title bar. Clicking on the Zone tab displays the Zone Editor in the space above Sampler (see Fig. 1). The left side shows a list of samples organized in layers, and the right side shows either the Key Zone Editor or the Velocity Zone Editor, depending on which button you press. Key Zones graphically display MIDI note assignments, and Velocity Zones determine the Velocity range to which a sample layer responds. Immediately above the graphical object representing each zone is another object representing its fade range, allowing you to specify crossfades between samples.

The Sample tab displays the currently selected sample's waveform and lets you edit its assigned characteristics, such as root key, loop points, and sustain and release modes. Looping parameters are especially flexible and allow sustain and release loops that play once, repeat forward, or alternate forward and back. You can specify parameters such as panning and detuning, or whether a sample plays in reverse. Also in the Sample tab, the RAM button globally determines whether multisamples are loaded entirely into RAM or stream from disk.

The Pitch/Osc tab displays settings for the pitch envelope and for the modulation oscillator, which generates various waveforms for amplitude- or frequency-modulating the multisample. The Filter/Global tab provides access to Sampler's resonant multimode filter, which offers 12 and 24 dB-per-octave morphing filter types that shift from one response to another and back again. Also available in Filter/Global are the Shaper, which imparts waveshaping distortion, and loopable ADSR envelopes for the filter and amplifier.

FIG. 1: Ableton Live 6 features an optional software instrument called Sampler. With ample layering, sound-shaping, and modulation options, Sampler is surprisingly powerful.

The Modulation tab displays modulation sources that include another loopable EG and three LFOs. You can assign one LFO to modulate pitch, volume, pan, and filter frequency; the other three sources provide menus to select two routings to 25 destinations. Destinations include unusual selections such as sample offset, loop length, and filter morph. You can even route modulators to modulate modulators. You'll find additional mod routings in Sampler's MIDI tab, which lets you assign MIDI Note, Velocity, Release Velocity, Aftertouch, Mod Wheel, and Pitch Bend to modulate two parameters each, selected from the same list of 25 destinations.

Sampler imports sample libraries in Akai S1000 and S3000, EXS, Giga, Kontakt, and SoundFont formats. For most, it copies the sample data into its library, but for EXS and Kontakt multisamples, it simply links to the original AIFF or WAV files unless you choose otherwise.

If you purchase Live on disc, it includes SoniVox's Essential Instrument Collection (EIC), which furnishes about 14 GB of sampled content (and if you download Live, you can purchase EIC separately for $119). Instrument categories include all the usual suspects, such as guitars, drums, and orchestral instruments, as well as electronic textures and a 3.5 GB grand piano. When you load most instruments, they'll open in Live's basic sample player, Simpler. To put them in Sampler so that you can access more parameters, just right-click on the Simpler title bar and select Simpler→Sampler from the contextual menu. When you save your changes, the instrument will be saved as a Sampler preset.

## MANUFACTURER CONTACTS

**Ableton** www.ableton.com
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**Cakewalk** www.cakewalk.com
**Digidesign** www.digidesign.com
**E-mu** www.emu.com
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### Apple EXS24 mkII (Mac)

Emagic first launched EXS24 in 2000, making it one of the mature products in this lineup; only GigaStudio has been around longer. First as an optional add-on to Emagic Logic Audio and later as an integral part of Apple Logic Pro, EXS24 has established its native format as one of a handful of de facto standards. The EXS24 mkII plug-in runs only as an Audio Instrument object within Logic Pro 7 ($999); there is no standalone version. It runs native on any Macintosh that can run Logic Pro, and it runs as a TDM plug-in on TDM-based systems courtesy of Emagic System Bridge (ESB) TDM. It can stream 24-bit, 96 kHz samples direct from hard disk and import a variety of sampler formats. EXS24's polyphony maxes out at 64 voices, and though it is not multitimbral, you can run as many as 64 instances if your system can handle it. Logic Pro and all its instrument plug-ins, including EXS24, are copy protected and require an Apple USB dongle to operate.

EXS24's main interface contains all the plug-in's knobs, sliders, and displays and provides access to its sound-shaping features (see **Fig. 2**). To load a program, just select it from the pull-down Sampler Instruments menu just above the prominent filter section. Instruments are arranged hierarchically in folders. Most of the sliders are split into upper and lower halves, allowing you to specify a range of values. For example, using the Level slider, you can control the minimum and maximum output produced by low and high Velocities. In addition, Logic Pro lets you automate any changes you make.

The filter offers six responses: 12 dB per octave highpass and bandpass, and 6, 12, 18, and 24 dB per octave lowpass. The Fat button boosts the bass when you increase the lowpass filter's resonance, and a Drive knob imparts warm distortion. If you don't need the filter, you can turn it off to save CPU cycles.

In the GUI's center is the modulation matrix, which offers ten modulation paths, each connecting one of 22 destinations with one or two sources. If you select two sources, the value slider will split and allow you to set a range. One source can control many destinations, and many sources can control one destination. The two ADSR generators and three 7-waveform LFOs are freely assignable mod sources. The envelopes have an additional Hold stage that you can set only in the Instrument Editor.

Clicking on the Edit button opens the Instrument Editor, which allows you to assign samples and multisamples to zones, place zones in groups, and organize groups into Sample Instruments. Within each zone, you can specify a sample's note range, start and end frames, loop points, and other parameters. Buttons in the loop and start- and end-frame sections open Logic's Sample Editor, which displays the sample's audio waveform. There you can graphically adjust loop points and other parameters, as well as perform various waveform edit functions such as copy and paste, normalize, adjust tempo, convert sampling rates, and so on.

Clicking on the Options button reveals a menu for specifying preferences, importing foreign sampler formats, and saving and reloading instruments and settings. EXS24 imports Giga, SoundFont 2.0, SampleCell, ReCycle, and Akai S1000 and S3000 sample and instrument formats. Conversion is generally automatic and requires only that you select the files to be imported.

Logic Pro 7 comes with about 3.5 GB of sampled content that comprises mostly bread-and-butter instruments such as piano and drums, as well as orchestral instruments and a large selection of synth timbres. Because EXS24 also opens GarageBand instruments, any installed on your computer will appear in the Sampler Instruments menu.

### Cakewalk DS864 (Win)

DS864 is one of several instruments that are part of Project5 version 2.5 ($259), a digital audio sequencer bundled with numerous soft synth and effects plug-ins. Once you install Project5, you can use DS864 in any host that supports DXi plug-ins. DS864 is a 64-voice polyphonic, 8-part multitimbral sampler that allows you to import and graphically map 16- or 24-bit samples in WAV or AIFF format and save multisampled instruments in its [illegible] DP8 format. It comes with 500 MB of sampler programs and banks in DP8 format from Q-Up Arts. Programs range from acoustic and electric pianos and basses to orchestral instruments and synths.

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parts, each assigned its own MIDI channel and audio outputs. Each part can be a different sampler program, and groups of as many as 128 programs are stored in banks. You can map note and Velocity ranges for each sample, and shape sounds using traditional techniques such as filtering and envelope modulation.

DS864's graphical mapper dominates its left side (see **Fig. 3**). When you add a sample, you can click on a tiny keyboard to set the root pitch and click-and-drag a rectangle in the display to define key and Velocity ranges. You can't enter mapping data numerically, but you can use onscreen knobs to adjust tuning, gain, panning, and key tracking. You can also invert the sample's phase and play it in reverse. Although DS864 does not offer the ability to define loop points (you'll need a separate waveform editor for that), you can choose to play loops either forward or alternating forward and reverse.

On the right side, DS864 has two filters; one has a fixed 12 dB-per-octave slope with resonant lowpass, highpass, bandpass, and notch responses, and the other is a 24 dB-per-octave lowpass filter with self-oscillating resonance. They can be arranged in parallel or in series, and each offers three modulation sources: an envelope generator, an LFO, and keyboard tracking.

Four EGs are dedicated to controlling the filters, amplitude, and pitch. All four are identical and go far beyond ADSR by providing eight stages and looping. Their modulation depth responds to Velocity and Aftertouch. Three especially versatile LFOs can modulate pitch, amplitude, and panning. In addition, one can modulate the multimode filter's frequency or resonance, and another can modulate the other filter's frequency or resonance. Along with knobs for controlling rate and depth, each gives you a choice of five waveforms, four trigger modes, and a knob to set the initial phase.

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FIG. 3: DS864 is just one of the DirectX instruments bundled in Cakewalk's Project5. Its entire GUI appears in a single window.


### Digidesign Structure (Mac/Win)

Digidesign has introduced an RTAS sampler plug-in that is so new, it is still in public beta as of this writing. Structure (price TBD) is compatible with all versions of Pro Tools 7 and later, and like Pro Tools, it's copy protected using an iLok USB dongle. The plug-in supports 24-bit audio at rates up to 192 kHz and surround formats up to 7.1. Though you can limit polyphony to as few as four voices, it is theoretically unlimited, and it is 128-channel multitimbral (if your computer is up to the challenge). Structure is not a standard RTAS plug-in, however, because it's closely tied to Pro Tools' audio engine, which manages the distribution of sampler voices and audio tracks.

Structure's user interface provides easy access to a variety of windows. On the main page, the patch list is on the left, the parameter panel is on the right, and the keyboard/smart-knob section is below that. The latter furnishes an 88-note onscreen keyboard, six reassignable knobs, the Master output knob, and a wide parameter display (see **Fig. 4**). The knobs are linked to whatever parameters you choose, and each can be linked to multiple parameters. Associated parameter names appear above each knob, and turning one shows its value in the display.

The main page lets you access parameters such as tuning, glide, and effects sends, depending on which tab you select. You can draw waveshapes or step patterns for the two LFOs, choose from six destinations, and set up randomization. The Effects page gives you five sends, each with slots for four global effects. Clicking on the effects processor's name displays its parameter controls. The browser lets you search your directories and display only certain file types, if desired, and the database lets you search previously scanned files.

In Structure parlance, a patch contains one or more parts, which are either multisampled instruments, insert effects, or MIDI effects modules. The parameter panel's content depends on what's selected in the patch list; if it's a patch, you can choose from four pages—Patch, Control, Mod, and Output—by clicking on tabs. If it's a part, you get five page tabs—Part, Filter, Amp, Mod, and Output. Selecting audio effects in the patch list displays the effects processor's control panel, and selecting a MIDI Module displays its parameters.

Structure's MIDI Modules serve more or less the same function as Native Instruments' Kontakt Script Processor or Tascam's iMIDI Rules Manager (which I'll discuss later). MIDI Modules change the way a part is played by randomly transposing controller values in

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FIG. 2: In addition to its main interface, Apple EXS24 mkII furnishes Sample and Instrument Editor windows for creating and modifying multisampled instruments.

In the GUI's center is the modulation matrix, which offers ten modulation paths, each connecting one of 22 destinations with one or two sources. If you select two sources, the value slider will split and allow you to set a range. One source can control many destinations, and many sources can control one destination. The two ADSR generators and three 7-waveform LFOs are freely assignable mod sources. The envelopes have an additional Hold stage that you can set only in the Instrument Editor.

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DS864's GUI comprises just one window, from which you can load and layer as many as eight multisampled

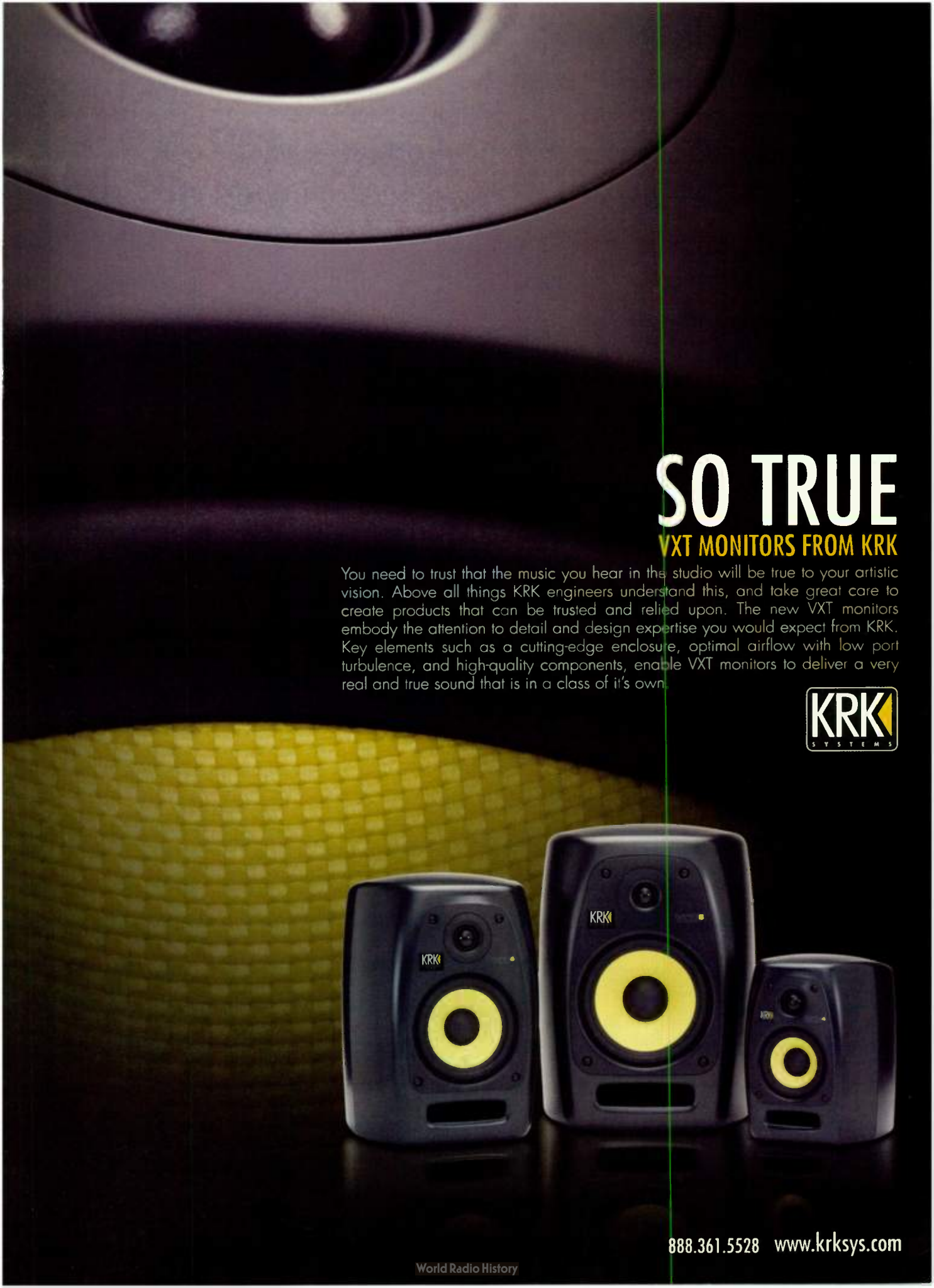

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FIG. 4: Digidesign Structure uses Pro Tools 7's audio engine to handle the distribution of sampler voices.

response to how long it's been since the previous Note On, for example, or by triggering a particular response when you play a chord.

In the patch list, each patch has an Edit button; clicking on it opens an Editor window you can enlarge to any size you like. That's where you modify sampler parts within a patch. It has Treeview, Mapping, and Edit sections. The Treeview section lists the parts and all the samples within them. The Mapping section displays the selected part's keyboard and Velocity assignments and lets you graphically modify them. The Edit section displays controls for sound-shaping parameters such as the filter and two EGs. The resonant filter offers 20 modes, and you can edit the 10-stage envelopes graphically by clicking-and-dragging breakpoints. If Wave is selected, the Mapping section displays the selected sample's waveform and allows you to define loops and crossfades and perform other edits.

Structure was designed for quickly creating your own sampled instruments, and the ability to drag-and-drop Pro Tools regions directly into Structure takes advantage of the tight integration between recording and sample playback. Because it's still in beta at this point, I can't tell you much about what content Digidesign will include except that it should be larger than 10 GB. Judging from the handful of patches that accompany the beta version, though, it's obvious that Structure's content makes heavy use of keyswitches to play alternate articulations. In addition, Structure can load EXS, SampleCell, and Kontakt (versions 1 and 2) files just as if they were its own file format. Additional formats such as Akai, E-mu, and Giga will be supported through a free conversion tool at some point after Structure's release date.

## E-mu Emulator X2 (Win)

In 1981 pioneering synth builder E-mu Systems unveiled the first dedicated sampler, the original Emulator. It was an 8-bit digital instrument that stored samples on 5.25-inch floppy disks. E-mu continued to build increasingly sophisticated hardware samplers until making the jump to software in 2004 with the introduction of Emulator X. The latest version, Emulator X2 ($399.95), is a 64-part multitimbral standalone application and a 16-part VSTi plug-in. It supports 24-bit sampling rates as high as 192 kHz, and it allows you to record samples directly into software using techniques familiar to hardware sampler users.

Emulator X2 comes with its own hardware, a 2-in/2-out USB MIDI interface called the XMIDI 2x2, which serves as a dongle and is required for the software to run. If you own an E-mu sound card or USB MIDI controller, however, those can serve as dongles too. An E-mu sound card offers the additional advantage of using PatchMix DSP to handle audio signal routing within Emulator, as well as DSP-accelerated effects.

The software is organized in a modular fashion and inherits some notable features from its hardware counterparts, including Z-Plane morphing filters and a hierarchical data structure with its roots in EOS (Emulator Operating System). Emulator's fixed-size GUI has two panes and a toolbar across the top, and you can optionally display a free-floating keyboard (see Fig. 5). The pane on the left contains the Tree view, a browser that enables you to locate sample content. When you open a bank, the Multisetup page appears on the right. It contains 64 slots (4 groups of 16, selected with tabs), each with settings for volume, panning, effects, and output assignment. You can select one preset for each slot and link MIDI Control Changes to various parameters.

Typically, you'll save a Multisetup with all the presets you'll need in a song or sequence. A preset contains one or more voices assigned to respond to a single MIDI channel. A voice is a sample or a multisample assigned to keyboard and Velocity zones. Voices and Zones pages

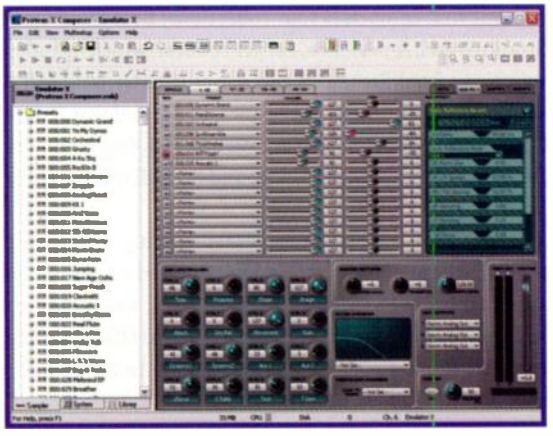


FIG. 5: Replacing generations of sampling hardware from E-mu, Emulator X2 is one of the few software instruments available that can actually record samples.

allow you to adjust voice parameters such as levels and tuning. The Voice Processing window offers sound-shaping sections such as a single multimode filter and modulators.

Fifty-five filter types range from 2-pole lowpass to Morph Designer, a user-programmable filter with up to six stages. Modulator tabs let you select from three 6-stage EGs, two 17-waveform LFOs, a lag processor, and three FM generators. Voice Processing's Cords section lets you connect 95 modulation sources to 105 destinations that include two lag processors, three function generators, and even other mod routings.

Emulator has a full complement of effects that can be part of a preset or a Multisetup. Each has plenty of user parameters and flexible routing. All the classic effects types are available, from reverb, chorus, and compression to parametric EQ, tube simulation, and beat-synced tremolo.

Emulator X2 has several additional features I should mention. One of these is TwistaLoop, a beat-detection function that analyzes a sample's rhythmic components in the same manner as time-slicing, and helps to locate and select loop points and regions. It also improves the quality of time compression and expansion and enables you to continuously control a loop's playback speed and choose which loops play and repeat in real time. TwistaLoop turns Emulator into a versatile groove sampler.

Another feature that other samplers lack (though third-party utilities offer similar functionality) is SynthSwipe, which automatically samples MIDI instruments and creates finished Emulator presets. After you set up parameters such as input level, note range, interval, Velocities, and durations (key up, key down, and pause), just click on Record, wait for completion, and save your new bank and samples. Emulator X2 can also resample its own output.

Emulator X2 ships with more than 3 GB of sample content that comprises four collections: Beat Shop One, Studio Grand, X-perience, and X Producer, which includes selections from Proteus X and a complete General MIDI sound set. Additionally, the included Emulator X File Converter application can convert Akai, E-mu Emulator III and ESi, EXS, Giga, HALion, and SoundFont files into Emulator's native EXB format.

Emulator X2 Platinum, slated to ship soon after this article appears, will include 32-bit and 64-bit native applications for Windows XP and Vista, the ability to stream from hard disk, and more than 20 GB of sample content. Emulator X 2.5, a free upgrade to X2 owners, should follow a few months later.

### MOTU MachFive 2 (Mac/Win)

Still in beta as I'm writing this, MachFive 2 ($395) is the successor to MOTU's sampler plug-in, which first

FIG. 6: MachFive 2 is the newest version of MOTU's versatile sampler. Adding to the previous version's features, it runs standalone, offers rule-based layer switching, and includes a 32 GB sound library.

shipped in 2003. Version 2 adds standalone operation, a modular synthesis engine, and more. It supports AU, DXi, MAS, RTAS, and VST plug-in formats, and it opens most foreign sampler formats without the need to convert them to its native M5P format. MachFive allows you to drag-and-drop audio directly from Digital Performer and other supported hosts. It has extensive loop-editing and beat-slicing capabilities and full-screen waveform editing with unlimited undo and redo. It even offers a built-in tuner and real-time spectrum analysis.

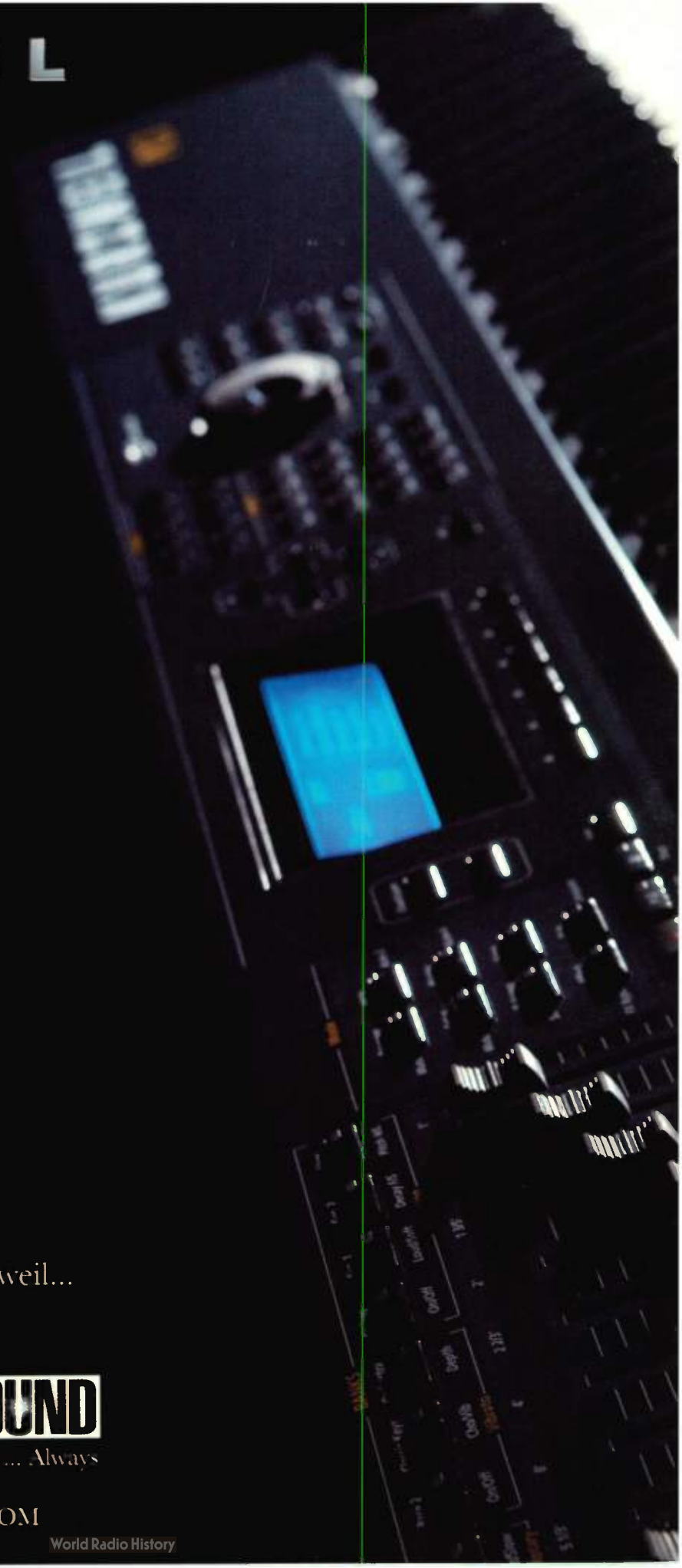
MachFive supports 24-bit, 192 kHz audio and surround formats up to 7.1. Theoretically, it can play an unlimited number of multitimbral parts, and you can even specify which parts stream from disk. Unless you disable streaming, you can specify how much sample data will be preloaded into RAM. Polyphony is also unlimited, and it can receive on 256 MIDI channels. Like Digidesign Structure, MachFive uses an iLok USB key for copy protection.

Most of MachFive's GUI is in a single window, which is divided into sections such as a browser, parts list, display area, and so on (see Fig. 6). MachFive makes good use of contextual menus and windows you summon by right-clicking; right-clicking on almost any knob, for example, opens an Automations and Modulation window for defining control sources.

You can view eight parts at a time, each containing a preset; scrolling reveals additional parts. You can assign instruments to some parts and loops to others, and then save all the parts collectively as a performance. You can assign as many parts as you like to the same MIDI channel. MachFive's Expert mode opens a window for defining key ranges, Velocity zones, and keyswitches. You can create any number of layers within a preset and apply MIDI rules for switching between them.

MachFive's Mapping Editor lets you create voices

# KURZWEIL



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Elton John
Paul McCartney
Mike Garson
The RZA
Greg Mathieson
Alicia Keys
Kenny Rogers
Stevie Wonder
John Carin
Rick Wright
Jordan Rudess
Steve Walsh
The Boston Pops
The Dresden Dolls
Peter Gabriel
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from scratch using the Create Synth command and then layer them with recorded samples. Raw Oscillators lets you stack as many virtual analog oscillators as you like, each with a choice of five waveforms and individual gain, pan, and tuning. Organ Emulator provides nine onscreen drawbars for specifying harmonics, as well as percussion and rotating-speaker parameters.

Each of MachFive's two filters lets you choose from eight filter types, including lowpass, highpass, bandpass, notch, and comb. A filter topology menu determines the filter routing in series with the effects and overdrive. Six loopable EGs can be either AHDSR (ADSR with an additional Hold stage) or Multi; a Multi envelope can have any number of breakpoints. An Edit button in the EG section opens a large window for editing envelopes, and you can save and reuse any Multis you create. In addition, you can modulate the amplifier, pitch, or filter frequency with any of 19 sources.

Two buttons near the display area let you toggle MachFive's Mixer view and waveform editor, and two others let you expand the waveform editor to fill most of MachFive's GUI or open it in a separate, full-screen window for detailed editing. The editor offers Sample, Stretch, and Slice modes. In Slice mode, you can enable MachFive's Loop Lab, in which you can define, stretch, rearrange, and map beat slices manually or automatically.

MachFive has a long list of multichannel effects processors with hundreds of presets, and you can save user presets. Effects include a simulated tape delay, a talkbox filter, an 8-band EQ, a 3-band compressor, and a convolution reverb with a large selection of impulse responses. You can link four effects and save them as a group, and you can apply as many as four insert and four part effects for each part simultaneously with four aux and four master effects. You can also apply effects and effects groups to individual samples and key groups. All time-based effects can be synced to tempo.

MachFive can now open foreign formats directly—even CDs for Akai, E-mu, Kurzweil, and Roland samplers—with no translation needed. It opens EXS, Giga, Kontakt, SampleCell, and SoundFont files, as well as loops in Acid, REX2, and Apple Loops format.

MachFive 2 comes with a whopping 32 GB of content that includes assorted loops and phrases, synths and ethnic instruments, vocals and sound effects, an 8 GB grand piano, and orchestral instruments from Vienna Symphonic Library. The bundle also supplies a handful of instruments and drum loops sampled in surround and at high sampling rates.

### Native Instruments Kontakt 2.2 (Mac/Win)

Since its introduction in 2002, Kontakt has been Native Instruments' premier sampler. It is a semimodular instrument that runs standalone and as an AU, DirectX, RTAS, or VST instrument plug-in. In 2005, Kontakt 2 ($449) added some significant features, such as a new file format, convolution effects, 16-channel surround, and MIDI script processing. Kontakt is 64-part multitimbral and supports 32-bit sampling rates up to 192 kHz. It translates a dizzying variety of file formats, and its assortment of surround-enabled effects is quite impressive. Like other software from Native Instruments, Kontakt uses an automated online challenge-and-response copy-protection system.

Kontakt incorporates six methods of audio playback. The traditional Sampler mode loads data into RAM and changes duration when it shifts pitch. DFD mode is identical except that it streams data direct from disk and therefore cannot play samples in reverse. Tone Machine employs granular resynthesis to change pitch without affecting duration. Time Machine uses granular synthesis to shift pitch and stretch time. Time Machine II performs the same functions but is optimized for higher-quality transposition and time-stretching. Finally, Beat Machine specializes in creating and rearranging beat-sliced sampled instruments in the Loop Editor.

Kontakt appears onscreen as a rack of gear in a resizable window (see **Fig. 7**). It can fill your computer display or shrink to the size of a piano key. You can choose to show or hide its various editors, browser, keyboard, mixer, modulators, effects, and other portions of its GUI. On the left is the browser, which lets you view not only the contents of your sample library, but also every audio file on all your disks. The browser's database functions keep track of sample locations, and you can easily load programs or Multis into the main rack on the right. It can also load entire banks of programs and quickly switch between them in response to MIDI Program Changes. In addition, the browser displays effects and modulators you can drag-and-drop into the rack.

Kontakt's front panel doesn't display synthesis parameters like EGs, LFOs, and filters in the usual


**FIG. 7: Native Instruments Kontakt 2 combines a flexible sampling engine, extensive programming capabilities, and versatile file translation.**

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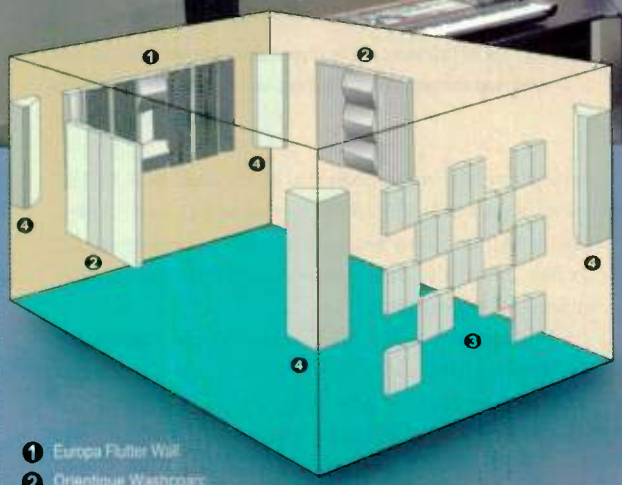


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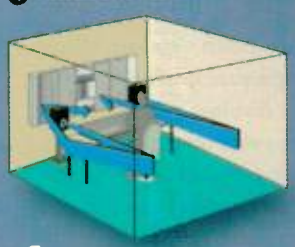
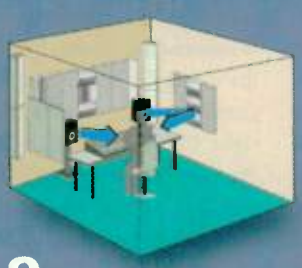


**1**

### Europa Flutter Wall

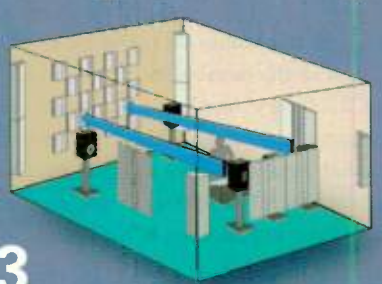
Reduces 'front-to-back' echo, standing waves and monitor foldback. Dozens of creative patterns to choose from. Prices start at $189**.



**2**

### Orientique Washboard

Absorbs primary reflections and side-to-side flutter. Eight creative patterns to choose from. Priced from $189**.



**3**

### Scandia Scatter Blocks

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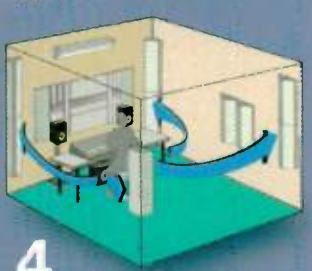

**4**

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manner. Instead, it supplies modulators you can add as needed. Modulation sources range from six LFO types and three EG types to a step modulator that lets you draw custom controller curves. Envelopes are AHDSR, DBD (two sections with a center breakpoint, suitable for modulating pitch), and Flexible, which allows you to create repeatable 32-stage controller curves.

Filters are categorized as effects, and you can choose from 13 types, from 1-pole lowpass to a CPU-intensive multimode filter that combines 3 stereo 2-pole filters in diverse configurations. Other effects include saturation, chorus, reverb, stereo simulation, modeled tube and transistor distortion, and a rather flexible convolution processor.

Kontakt's Loop Editor allows you to graphically define as many as eight loops for each sample. Likewise, the Mapping Editor lets you assign samples to notes and Velocity ranges, and it offers a choice of techniques for creating and editing zones. You can leave either Editor window in the rack or detach it as a separate window. Working with both windows open is especially convenient, because selecting a zone in the Mapping Editor displays it in the Loop Editor.

One feature that expands Kontakt's programmability is the Kontakt Script Processor (KSP). Kontakt furnishes a collection of ready-to-use MIDI-processing routines and allows you to create your own. The Script Editor lets you access scripted control panels and, unless they're password protected, edit existing scripts. The Kontakt 2 Script Library offers functions such as MIDI echo, automatic harmonization, arpeggiation, harp glissandos, and much more.

Kontakt claims compatibility not only with software formats such as EXS, Giga, HALion, MachFive, NN-XT, REX, SampleCell, and Bitheadz Unity, but also with sound libraries for hardware samplers, including most Akai, E-mu, Kurzweil, and Roland samplers and even Ensoniq EPS and ASR-10. It comes standard with a well-rounded 15 GB sample library that covers a lot of territory. In addition to the usual guitar, bass, and drums, the Kontakt 2 Library furnishes surround-specific synths and organs, beat-sliced loops, and KSP-scripted instruments that possess unique capabilities. Half the sampled content comprises strings, brass, woodwinds, and percussion from Vienna Symphonic Library.

### Propellerhead Reason NN-XT 3.0 (Mac/Win)

Just as EXS24 mkII is an integral part of Logic Pro, NN-XT is one of many instruments supplied by Propellerhead Reason ($499). Most Reason users don't look beyond NN-XT's functionality as a sophisticated sample player, but it's a very capable sampler that can load WAV and AIFF files and organize them into multisampled patches in its own SXT format. Another sampler in Reason is NN19, and rather than cover them both, I've chosen to write about the more powerful of the two. NN-XT can open patches in NN19's SMP format as well as REX files, which are native to yet another Reason instrument, Dr:Rex. It can also open presets and individual samples in SoundFont format.


FIG. 8: One of two samplers in Propellerhead Reason, NN-XT lets you map samples, apply filtering, and route modulators in its Remote Editor window.

When you open NN-XT, you see only the main panel containing three controller wheels, a few global controls, and an area for selecting patches. Alongside the pitch and mod wheels, a third wheel sends and responds to your choice of Aftertouch, Expression, and Breath Controller messages. The global controls furnish knobs for filter frequency and resonance, modulation decay, and amplitude attack, decay, and release.

Opening NN-XT's Remote Editor reveals all its additional parameters in a single panel, most of which is dominated by the Key Map display (see Fig. 8). In the display, a list of samples appears on the left with a graphic representation of their assigned zones on the right. Clicking on either a sample or a zone selects it and displays the sample's bit depth, sampling rate, and file size. You can also select a zone by playing a MIDI note, if you enable that function. If you Shift-click to select noncontiguous zones, any changes you make will be applied to all the selected zones.

A row of knobs below the display accesses 15 parameters for each zone, with values displayed just above the knobs. Right-clicking on a zone reveals a contextual menu for editing and sorting zones, creating Velocity crossfades, and performing other functions. An Alternate function determines the degree of randomness for selecting between two zones during playback.

NN-XT's multimode filter has three lowpass slopes and highpass, bandpass, and notch responses. One AHDSR generator controls amplitude, and another is available for modulating filter frequency and pitch. The

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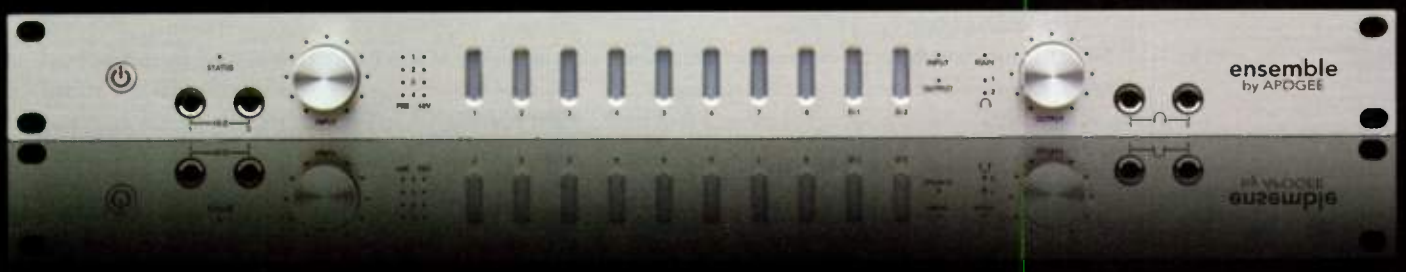


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Modulation section provides routings to control filter cutoff and resonance, LFO amount and rate, and modulation envelope depth and decay. One multiple-waveform LFO can modulate pitch, filter, and amplitude, and a simpler LFO modulates vibrato and pan position. On NN-XT's rear panel are inputs that allow you to modulate various parameters with other Reason instruments and devices such as Matrix and Redrum.

Most Reason data is in the form of ReFills, a compressed storage format containing patches, samples, REX files, SoundFonts, and demo songs. Reason 3.0 comes standard with two ReFills, Orkester and Factory Sound Bank, totaling about 1.25 GB. Exactly how much of that comprises NN-XT patches and samples is hard to determine, but it's enough to keep you busy with NN-XT for quite a while. You can purchase additional ReFills from Propellerhead and third-party soundware developers. Reason requires a serial number to run and occasionally requests that you insert your original installation discs as a copy-protection measure.

## Steinberg HALion 3.22 (Mac/Win)

HALion ($399.99) has been around since late 2001, and many sample libraries are available in its native format. The current version is 256-voice polyphonic and 16-part multitimbral, and it runs standalone, as a ReWire slave, or as an instrument plug-in for AU, DirectX, or VST hosts. It can import a variety of file formats and supports 5.1 surround and sampling rates as high as 384 kHz.

Other than GigaStudio, HALion was only the second sampler that could play samples direct from disk; now you can even specify how much disk streaming occurs, with settings from Very Very Low to High. It was also the first to let you recover RAM from a multisampled instrument using a function called RAMSave. After you've loaded an instrument and used it in a sequence, RAMSave actually keeps track of which samples the sequence triggers and deletes any that aren't needed.


FIG. 9: Steinberg HALion is a multitimbral sampler with a variety of views. Its RAMSave feature deletes unneeded samples from memory.

HALion's GUI lets you switch between seven main pages: Global, Keyzone, Loop, Sound, Browser, Options, and Macro (see Fig. 9). In any page other than Macro, the circular Pitch/Modulation Controller dominates the lower-right corner. You'll probably spend most of your time in Macro, however, which affords access to the controls you'll use most frequently. You can select programs, play an onscreen keyboard, and adjust parameters affecting the filter, amplifier, LFOs, envelopes, tuning, glide, and so on. In the Macro page's center, you can choose from five views: Quick Controls, for hands-on access to any eight user-specified parameters; Global, for selecting from 16 programs; Keyzone, for an overview of zone assignments; and Program and Send Effects, for assigning insert and global processors.

Also in the Macro page is HALion's resonant multimode filter, which has a dedicated ADSR and a choice of ten filter types, six of which model filters in classic Waldorf synthesizers. Another ADSR modulates the amplifier, and you have fine control over the tuning of each multisampled instrument. Two LFOs offer ten waveform types. You'll find more comprehensive control in the Sound page, which lets you graphically position and add envelope curve points, define a stepped modulation envelope, add saturation using the filter's Fatness knob, and so on.

In the Global page, you can load and select 16 programs and change their volume, panning, audio output, transposition, key and Velocity range, and Velocity curve, as well as route insert and global effects. The Keyzone page gives you complete graphical control of sample mapping, layering, and Velocity scaling. The Loop page furnishes a detailed waveform display in which you can define start and end points for a sustain loop and a release loop. Although the Loop page is specifically for editing individual samples, you can apply most editing operations in all pages either to selected samples or to all samples in a program.

The Options page contains global settings that affect file loading, memory usage, resampling quality, audio outputs, and MIDI controllers. In the Browser page, you can locate HALion content, organize directories, assign program categories, audition samples, copy contents from removable media, and import samples and programs from other file formats. HALion imports Akai, E-mu, EXS, Giga, Kontakt (version 1), Kurzweil, Roland, and SoundFont 2.0 samples, as well as groove files from Zero-X.

HALion 3 is bundled with 2.5 GB of original content and more than 1.5 GB of soundware demos from developers such as e-Lab and Scarbee. In addition to a nice collection of orchestral instruments, you'll find an ample number of drum kits, synths, pianos, guitars, chromatic percussion instruments, and more. HALion's copy protection is by means of a Syncrosoft-compatible USB dongle.

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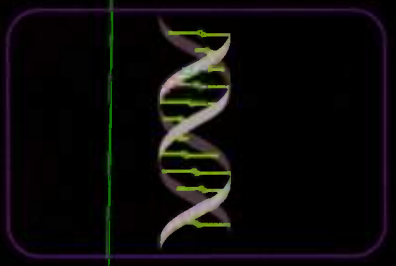

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### Tascam GigaStudio Orchestra 3.21 (Win)

Introduced in 2000, GigaStudio has been around longer than any other program surveyed here; its predecessor, GigaSampler, was first unveiled in 1998. Both programs were the first (and for years, the only) software instruments that could stream samples direct from disk in real time. They pioneered the concept of keyswitching, a practice that has become standard for instantly selecting different articulations of sampled instruments. Unlike most of the instruments surveyed here, GigaStudio is a true sampler in that it can record instruments directly, and its editing capabilities are quite deep. Thanks to GigaStudio's long-standing stature among audio professionals, a large selection of sample libraries in Giga format is available from third-party soundware developers.

The current version, GigaStudio Orchestra 3 ($599), has theoretically unlimited polyphony (limited only by hard-drive and processor speed) and 128-MIDI-channel reception. It can host VST effects plug-ins (in addition to its own NFX format) and can operate as a ReWire client. It plays and records 24-bit audio at rates up to 96 kHz, and it can record as many as 64 live audio inputs simultaneously. You can link GigaStudio to other programs on your computer—a sequencer, an audio waveform editor, and GigaStudio Instrument Editor—and open them with Quick Launch buttons in GigaStudio's toolbar. Also in the toolbar is the Audio Capture tool used for recording samples.

FIG. 10: Tascam GigaStudio Orchestra 3 is the latest version of the sampler that introduced hard-disk streaming and keyswitching. It delivers impressive programmability and 17 GB of content.

GigaStudio's GUI is divided into panes you can view, resize, or hide at will (see Fig. 10). The QuickSound Loader pane serves as a browser for locating, auditioning, and loading instruments, performances, samples, and impulse responses. The MIDI Mixer pane provides 16 slots, one for each MIDI channel. Port tabs at the bottom of MIDI Mixer let you select MIDI ports for 8 groups of 16 MIDI channels. You can assign multiple instruments to the same MIDI channel by clicking on the Stack Instruments On Active MIDI Channel button in QuickSound Loader.

Whereas GigaStudio Instrument Editor is a separate application for creating instruments and saving changes permanently, QuickEdit is for editing instruments nondestructively and saving them as part of an instrument or performance file. A button on each slot in MIDI Mixer opens the QuickEdit window, which comprises four sections: Keyboard, Articulation, Wave, and Dimensions. Pressing a note on the Keyboard selects the sample assigned to that note and highlights its key zone. The Articulation section lets you shape the selected sample by accessing GigaStudio's multimode filter (with a curve display), three EGs (a 2-stage and two 6-stage), and sine-wave LFO. The Wave section displays the selected sample's waveform and superimposes modulation curves such as EGs or LFOs, which you can modify graphically. Dimensions displays real-time control sources such as Velocity splits, sustain pedal, keyswitches, and so on.

When you open the DSP Station pane, it replaces MIDI Mixer and more closely resembles a real mixing console. It provides the means to mix, route, and process audio. Clicking at the top of a channel opens a Wide Channel view, in which you can assign insert effects and aux buses, set up dynamics and EQ, and control the stereo image.

Another feature I should mention is Intelligent MIDI (iMIDI). It applies performance algorithms that enhance realism by automatically alternating sounds during repetitive passages, allowing you to play more authentic legato lines and so on. You can assign iMIDI routines to instruments using the iMIDI Rules Manager, which allows you to add, initialize, and edit rules that control supplementary functions.

GigaPulse is Tascam's integrated convolution reverb processor. It uses impulse responses to model acoustic spaces, microphones, and instrument resonances. Operating as an NFX plug-in, it allows you to position mics within virtual space on a Placement Selection Grid. GigaPulse includes a large number of impulse responses, and the Pro version allows you to import your own. GigaStudio comes with four additional NFX plug-ins: reverb, chorus/flanger, tap delay/auto pan, and EQ.

GigaStudio comes with the GigaPulse Pro convolution reverb and a 17 GB sample library that includes orchestral samples from Vienna Symphonic Library and

GigaPulse impulses from Notre Dame de Budapest. Giga Virtual Instrument (GVI) has effectively replaced two previous versions, GigaStudio Ensemble and Solo. On the horizon, Tascam plans to ship a major upgrade later this year; GigaStudio 4 will support Windows Vista 32, Vista 64, and XP64.

### Yellow Tools Independence 1.5 (Mac/Win)

Although Independence ($499) first shipped at the end of 2005, it is still relatively unfamiliar to many audio professionals, despite its respectable sampling power. It features a multiple-page user interface, extensive customization features, and a sizable collection of effects. Independence runs standalone and as a plug-in for AU, DXi, RTAS, and VST hosts. It supports surround formats up to 7.1 and features a built-in multitrack mixer.

Independence divides its GUI into two main sections, basics on the left and working areas on the right (see **Fig. 11**). Basics include sections for loading instruments, adding and removing layers, and changing project, layer, and MIDI parameters. Clicking to load an instrument into a layer lets you select from a menu divided into categories. Each layer is assigned to a MIDI channel, and you can assign an unlimited number of layers to the same channel.

The parameters that appear in the working area depend on which button is selected across the top; each button selects a different editor. Quick Edit lets you access parameters such as volume, panning, tuning, filter, and effects. The filter section lets you select from ten lowpass, highpass, and bandpass presets or create your own. Remarkably, slopes vary from 12 to 72 dB per octave.

Another view, the Modules Editor, affords access to modulators and other modifiers. That's where you'll find an insert filter and effects, mod wheel and other controller routings, tempo-synced LFOs, graphic EGs, and a menu for specifying keyswitches. You can freely route mod sources to destinations and specify curves for the EG's time segments. Envelopes can be assigned to modulate volume, pitch, pan, filter frequency or resonance, and even the depth of other EGs. EGs come in AHDSR and unipolar or bipolar free varieties, and you can add as many breakpoints as desired to free envelopes.

FIG. 11: Yellow Tools Independence has a multiple-page GUI and an 18 GB collection of sample instruments. Owners can download a free utility that translates foreign sampler programs to its native format.

The Performance view lets you access Independence's flexible humanization features. You can set up as many as 32 variations for each sample, each of which will play whatever alternate sample you specify, either randomly or in order. Variations are useful for playing alternate hits on the same drum, for example, thus avoiding tedious repetition. The same page lets you set up Advanced Legato mode for assigning numerous parameters to individual samples to simulate legato playing techniques.

The Mapping Editor allows you to graphically create custom note and Velocity zones in a zoomable display. Clicking on a zone opens a waveform-editing window in which you can define start and end points, loop points, fades, and crossfades. It also displays various parameter values for the selected zone. Right-clicking in the window plays the selected sample. Additionally, the Mapping Editor features automatic groove recognition, and you can switch to Slice-Edit mode in the waveform display and adjust sensitivity for auto groove recognition. It can also import MIDI files and allows you to edit them as if they were keymaps. Independence can even play several MIDI files at the same time.

Yet another working-area view is the Mixer, which provides individual channel strips for each instrument and unlimited buses. In addition to the usual mixer parameters, each channel has inserts with an unlimited number of effects. Dozens of effects range from chorus, bit reduction, and additional multimode filters to a 6-band parametric EQ, mic and preamp modelers, and a convolution processor called Origami.

Independence is bundled with an 18 GB sound library that includes orchestral instruments from Kirk Hunter Studios, pipe organs from Notre Dame de Budapest, and many original sounds from Yellow Tools, a company that first made its mark as a soundware developer. Percussion, saxophone, and bass samples from Culture, Candy, and Majestic, respectively, are in abundant supply. You also get lots of guitars, pianos, and synthesizers, and Independence can open content from any current or future virtual instruments from Yellow Tools. For copy protection, Independence relies on a proprietary USB key ($40), which comes with software for managing authorizations.

## Superior Software

As you can see, your choices are legion if you're shopping for a sampler. Which one you choose, of course, will

depend on your needs and your budget. If you already own a substantial sample library, compatibility will no doubt be a deciding factor. You should carefully consider how much sound design you might want to do on your own, both now and in the future. The majority of samplers surveyed here will likely be around for a while, and you'd probably prefer software that will evolve as your need for a sampler grows.

If most of your work is in a sequencing environment, then a multitimbral sampler is usually easier to use than one that requires you to open numerous plugins whenever you want to play several instruments at the same time. A single instance of Emulator X2 can open 64 different instruments on as many channels, but you would have to open 64 different instances of, say, EXS24 mkII to achieve similar performance. Granted, the computer you own now may not handle 64 parts simultaneously, but nonetheless, multitimbral operation is very desirable.

For anyone who cut their teeth on hardware samplers, the advantages of being able to actually make recordings within samplers may be apparent. Emulator X2 and GigaStudio 3 have that ability, but programs such as Structure and Ableton Sampler provide the same functionality because they're so tightly linked to their hosts' recording engines. As soon as you record audio into a track, you can open it in your sampler and begin turning it into a playable instrument.

What does the future have in store? A few samplers already work a lot like multitrack recorders; sequencing programs will probably gain new features that offer and extend sampling functionality. It's possible that audio tracks and clips will eventually let you define note and Velocity ranges so you can trigger them with a MIDI keyboard. Expansive sample libraries are already becoming the norm, and several sampler makers have told me that their next update will include much larger libraries. Obviously, more powerful computers and larger, faster storage will eliminate concerns over resource conservation. In the future, samplers will offer deeper functionality, more and better effects, and truly unlimited polyphony. Software samplers already offer capabilities that make hardware samplers obsolete, and fortunately for us, that trend is bound to continue. **EM**

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*Geary Yelton has been experimenting with audio recording since the age of ten and working with computer audio since 1984.*

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JASON VAUGHN

# It's Only Make-Believe

By Mike Levine

Hollywood sound designer Scott Gershin reveals his tricks of the trade.

The sounds that you hear in movies, TV, and video games fit so seamlessly with the visuals that it's easy to forget just how much work goes into producing them. Whether the sound you're hearing is a bomb exploding, a spaceship taking off, or silverware clinking, you can bet that a lot of effort was put into designing and recording those sounds.

One of the preeminent sound designers working today is Scott Gershin, who has worked on many major motion pictures including *American Beauty*, the *Chronicles of Riddick*, and *Shrek* and a host of video games such as the *James Bond* and *Mechwarrior* series and *Lost Planet: Extreme Condition*. Gershin is the executive creative director of Soundelux Design Music Group, a postproduction sound company located in Hollywood, California. He is also the cofounder of the Interactive Entertainment Sound Developers, a new branch the Game Audio Network Group. I had the opportunity to speak with him recently about his work, his gear, and his techniques.

**Is the term *sound designer* a good way to describe what you do?**

Sure, but sound design has become a very generalized and overused description. There are music sound designers, theatre sound designers, movie sound designers, game sound designers, and dialogue designers. What I do is take a look at the story or the action that I'm dealing with, and [then] use and manipulate sound to help tell the story or enhance the experience of the movie goer or game player. My job is to use sounds, whether captured from real life or manufactured and manipulated, to push the emotional buttons of the audience.

**Are you always working with a visual component?**
Mostly.

**Our readers are interested in the nuts and bolts of how you create sounds. Let's take, for example, a film situation. Can you describe your work flow? That is, when do you start preparing, and when do you see materials or visuals?**
Every film is a little different, but in a nutshell, I'll receive a script or a rough version of the film. I'll go through the script or video and start looking at potential audio opportunities. Sometimes the movie is already being shot when I get the script; other times, it's still in preproduction. In the latter case, I'll try and sit down with the director and talk about audio opportunities during filming that we might be able to record. If he or she is going to be filming large crowds, weapons, special vehicles (armored attack vehicles, race cars, aircrafts, submarines, and so on) or something that's unique and hard to get access to, then I'll recommend we go out and record those sound sources during the filming, as well as [during] those days before, between, and after the filming of those scenes.

FIG. 2: Gershin uses a variety of field recorders, including the 4-track Sound Devices 744T, when gathering sonic material for his work.

**You're talking about recording at the set of the movie?**
Yes, I do it quite often. On *Herbie: Fully Loaded*, the second unit had full access to California Speedway filming the NASCAR race. I discussed this opportunity with the director and Disney, and they gave my crew and me full access. On one of the days, there was cloud cover at the speedway; they couldn't match the scene previously shot with the actors, so they had to wait while the weather cleared up. That gave us a chance to wire the cars up, race them around the track, and get some amazing recordings. Disney also built 30 different Herbies to use in different parts of the film. Each one had a slightly different engine setup, so I auditioned them and chose four that I thought had unique sounds. We miked them up and spent days just recording Volkswagen Herbies at an airfield we rented north of Los Angeles. We installed switches in the car that allowed us to turn different spark plugs on and off, giving us the ability to record a palette of sounds to help create Herbie's "emotions."


FIG. 1: Gershin (left) and Peter Zinda mic the avionics of a jet plane while gathering sonic material for the *Chronicles of Riddick*.

COURTESY SCOTT GERSHIN

**Which do you use for your remote recordings—field recorders or laptops?**
I'm using mostly field recorders. I've tried laptops, but because of battery issues and glare, I've found them a little inconvenient. Most of the time I need to be fast, flexible, and agile. When I go out to record, I use multiple recorders and a team of recordists covering different perspectives of the sound, similar to a multicamera shoot (see **Fig. 1**). We use a combination of Sound Devices 744s (see **Fig. 2**) and Fostex FR-1s. Sometimes we'll use a [Zarcom Audio] Deva V or a Nagra. Microphone choices depend on what we're recording; each mic has a different color, reach, and purpose. We choose the right mics for the right purpose, similar to tracking an album. I just bought a Sanken 5-channel microphone, which I've been quite impressed with. Between Soundelux, myself, and the guys I record with, we have a very impressive mic arsenal to choose from.

**Once you're done gathering raw material, where do you go to work on the sounds?**
The studio that I work out of most is at Soundelux, although I also have a setup at home. The Soundelux studio is a minitheater; it's about 30 feet long and 24 feet wide with a full screen in front. I'm surrounded by computers, outboard gear, controllers, and anything that can manipulate sound (see **Fig. 3**).

**Do you want the room to sound kind of like a theater does?**
I try to simulate as much as possible the environment that the sound is going to play back in. So I've got my

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film set of speakers, and I've also got another set of speakers for my video-game work.

**Are those other speakers studio monitors?**

Yeah, my mid/close-fields. I'm using JBL LSR32s with a Bryston amp.

I actually have four speaker setups that I use. I have my EAW theater speakers behind the screen, my JBL LSR32s on stands at my main studio, and a 5.1 Dynaudio Air system that I run digitally out of my Pro Tools setup in my second studio. For critical listening, I also use my home-theater setup in which I'm using Thiels and a Sunfire sub.

JASON VAUGHAN

FIG. 3: Gershin's main studio at Soundelux is set up in a circular work space inside a large room designed to resemble a movie theater acoustically.

**Do you work mainly in Digidesign Pro Tools?**

Yes. I originally started off with a Synclavier, then became the first Waveframe user in postproduction using the Waveframe 1000. And then I converted over to Pro Tools. One of the advantages to Pro Tools is the ability to interface relatively well with the Avid and dubbing stage systems. Pro tools is the de facto standard in the industry. There are other systems, but because of time restraints and flexibility around town, Pro Tools is a good solution.

**I assume there are times when you create sounds without any field-recorded material?**

Sure—with synths, [with] sounds I record on Foley stages, [and by] manipulating sounds that I have collected. The great thing about Soundelux is that we've got an ever-growing library; it never gets stale. It's a combination of what we have recorded and what we've collected. For example, tomorrow I'm going to be recording llamas. While doing a Podcast, which I do every other week at http://nowcastnetwork.com, I met a family of musicians on the show who raise llamas. They were gracious enough to allow me and [sound designer] Peter Zinda to spend the day with them recording their llamas. I'm not necessarily using the llamas for any specific show, but the opportunity came up and we found out that llamas have a great wealth of vocalizations. Eventually, I'll need them for some creature sounds or [will] use them to embellish something else.

**Can you describe how you typically work with a director on a film job?**

Every director likes to interface differently. But basically the goal is to meet with the director to interpret what his or her vision is for the film. Some of the directors are very audio savvy and some are not. So then it's my job and my taste—that's why they've hired me—to interpret that vision and come up with my take on what they're trying to create. And then I'll start playing back scenes with the director, getting a feel for what his or her tastes and likes are, and I'll form a creative relationship with that director and with the picture cutter [the film editor]. Sometimes the sounds I need to create are very simple. When I did *American Beauty*, one of the things the director wanted was for the sound to help embellish the story [without

## SCOTT GERSHIN: SELECTED CREDITS

Films:

*Underworld Evolution* (Screen Gems, 2006)
*Herbie Fully Loaded* (Walt Disney Pictures, 2005)
*The Chronicles of Riddick* (Universal Pictures, 2004)
*Team America: World Police* (Paramount Pictures, 2004)
*Blade II* (New Line Cinema, 2002)
*Shrek* (Dreamworks SKG, 2001)
*American Beauty* (Dreamworks SKG, 1999)
*Godzilla* (Tri-Star Pictures, 1998)
*Braveheart* (Paramount Pictures, 1995)
*JFK* (Warner Brothers Pictures, 1991)

Games:

*Lost Planet: Extreme Condition* (Capcom, 2007)
*Transformers: The Game* (Activision, 2007)
*Onimusha Dawn of Dreams* (Capcom, 2006)
*Need for Speed: Most Wanted* (EA Mobile, 2005)
*Devil May Cry 2* (Capcom, 2003)
*James Bond: Everything or Nothing* (Electronic Arts, 2003)
*James Bond: Nightfire* (Electronic Arts, 2002)
*Mechwarrior series* (various publishers, 1994-2002)

A full list of credits can be found at www.IMDB.com.

calling] attention to itself. In other words, to totally blend with the production while paying special attention to negative space—that is, using silence as a tool. I had to find a whole new way of recording Foley and Walla group [actors who speak dialog that's used as a background effect] to seamlessly blend with production. The bottom line is that my job is to make sure that nobody knows I exist. My job is to create illusions with audio. Because what we're really trying to do is tell a story.

**A lot of times you make things larger than life, sonically?**

So many times you hear the phrase *the Hollywood Sound*. It's when you embellish real-life sounds in a movie, because many times realism is kind of boring. For instance, the classic example is the Hollywood punch. If anyone hit somebody and produced that sound, it would be a one-punch fight. Like in *Herbie*, I needed to create a Volkswagen with personality. A Volkswagen by itself doesn't have much personality, let alone the ability to emote joy, sadness, and jealousy. So what I needed to do was what all sound designers strive for—to create the right balance between realism and fantasy.

**How did you do that with Herbie's personality?**

Recording the realistic Herbie and manipulating its engines and driving in unusual ways turned out to be really effective, but it wasn't enough to produce the range of emotions I needed for the movie. Over my career I've used my voice to help make a lot of inanimate objects and characters come to life, giving [them] some personality. So I processed my voice and kind of emulated what the engine was doing and created another level of sound that, when combined with the organic sound, gave it a bit of a personality. Same thing when Herbie smiles or frowns using his bumper—I ended up manipulating real metal from a junkyard, and then added my vocalizations so it had the effect that makes the audience feel like it's talking. But I had to be careful not to go too far; otherwise, it would have become cartoonlike.

**Can you take a sound and repitch it and completely change its character?**

Yes. These days, there are two ways that I deal with pitch. One, I use pitch plug-ins that have pitch-envelope capabilities, and the two that I like the most are Serato Pitch 'n Time and the Waves Sound Shifter. For other types of pitch-shifting, I use samplers. Lately, I've been using Native Instruments Kontakt. I'm very interested in seeing Digi's new sampler. I've used the Waveframe 1000, the Emulator 4, and the MOTU Mach Five, among others. Ultimately, its great if I can take a sound from a track that I've been manipulating and drag it into a sampler from Pro Tools. But the bottom line is that the sampler or plug-in has to sound good.

**What are the basic techniques that you typically use when you're manipulating a sound?**

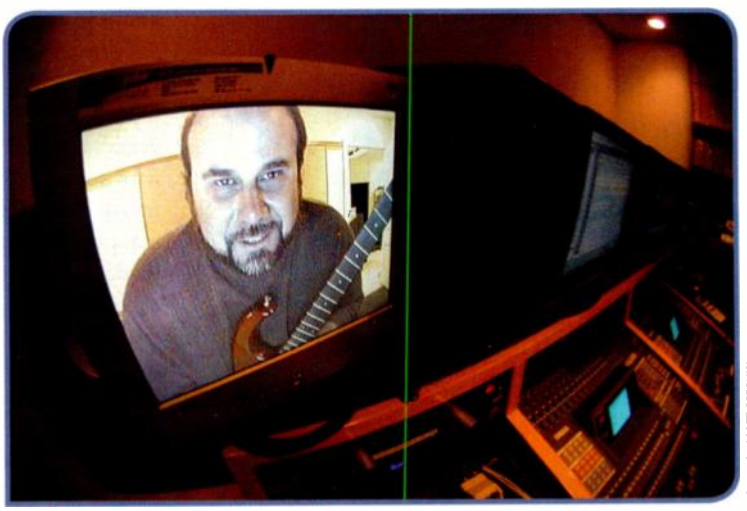
Pitch- and time manipulation is huge. I might pitch it and keep the speed the same, adjust the speed but keep the pitch the same, or adjust both pitch and speed. There are many ways to manipulate pitch and time [speed]. The other area I address is the tone using EQs and filters. You want to be able to reshape a sound and mangle it into something totally different. Another technique is to combine one sound against another and create something totally new. It could be as simple as grabbing five or six gun recordings, which, when combined and manipulated, get the results you're looking for. Or blending multiple animals together to come up with a new creature. Then there's obviously modulation-style and spatial effects that take a sound to a whole other level. Those can be used when you need it to be otherworldly—to sound like something that doesn't exist in nature.

**Can you give an example of a sound you created that would fit in the otherworldly category?**

When I did the *Chronicles of Riddick* (see **Fig. 4**), I used a lot of synthesizers and other instruments. I recorded a guitar through a full Marshall stack, loosened all of the strings, and then bent up the strings, combined with two or three harmonizers—all pitching up at the same time—to create the sound of the starship lifting off. Again, there's no right or wrong way. There's a lot of experimentation and trying to come up with ideas that may sound great or that sometimes aren't so great. Or totally wonderful accidents [can] occur that make you go, "Wow, this is really unique."

**What about the whole ambience issue? Do you use convolution reverbs with odd impulse responses like, say, the inside of a vacuum-cleaner tube or something like that?**

FIG. 4: Gershin, in screen, is shown with the guitar he used, with strings loosened, to create part of the starship-liftoff sound in the *Chronicles of Riddick*.



COURTESY SCOTT GERSHIN

Absolutely. [Audio Ease] Altiverb and [Waves] IR-1 are huge components. We've only scratched the surface of all the things that they can do. When you start doing convolution recordings in spaces, with materials, it becomes fascinating.

**What do you mean?**

Again, like a vacuum tube or something that you would never think you'd want from an acoustical space, but something that may manipulate the sound in very unique and odd ways. That's something I've been using for a couple of years now, again, [and] it's pretty much an amazing thing. I also still use a lot of classic reverbs—a lot of Lexicon and TC Electronic stuff. It really depends on what you're trying to create. With *Riddick*, I used musical instruments to create the core of the design, rather than going with stuff that's more predictable.

**So that it's a more original sound?**

Yeah, just something a little bit different. And again, we're kind of audio photographers—constantly going through life listening for sounds. For example, my washing machine at my house sits above a downstairs bathroom, and when the washing machine is on, it creates this interesting resonance in the bathroom. It feels like a starship or a submarine, or something to that effect.

**Do you carry a recorder around all the time in case you hear something you want to capture?**

I've got a couple of recorders that I could grab at a moment's notice. I did a submarine movie a couple of years back and recorded a Jacuzzi with spinning jets that I had in my house. And that was the key element to the sound of the torpedoes' blades. I bought an underwater microphone and recorded Doppler byes with it. And it's just basically a Jacuzzi jet, but against picture it gives off a different illusion.

**How about Foley artists—is it their job to put in the regular sounds, like the footsteps and door slams? And what's the difference between a Foley artist and a sound designer?**

The Foley artist, location recording, and manipulating sounds are all tools for the sound supervisor/sound designer to use in creating a sound track. When I create sound effects on the Foley stage with the Foley artists [or *Foley walkers*], I will take the results and combine them with the design to help add definition, detail, and many times something interesting and new. I will, for instance, create a really big explosion—I move the room. But let's say that during the explosion there's dirt that falls on metal. Or I want to crack something before the explosion. I want to enhance it with another level of detail. I'll do that on the Foley stage. Most of the time, the designer will create sounds and combine the artistry of the Foley walkers to add a level of detail.

**So how would you define the Foley artist's job?**

Foley artists and their recording engineers work in studios that are filled with everyday objects that they know will make specific noises. They also have a multitude of shoes and surfaces so that they can re-create sounds, movement, and footsteps in sync with the picture. In their bag of tricks, they know that if they grab, say, a Whippoorwill branch, that they can create *whooshes* that sound like a boomerang. And they've got an arsenal and a knowledge base to create sounds based on the junk that they've collected. For example, we've simulated snow by using baking powder to get the right crunch and texture.

**From a sound-design standpoint, how does working on a video game compare with working on a film. Is it similar or totally different?**

We're using similar artistries, but the release format is a bit different.

In film, I'm trying to create a blend of sounds as part of the storytelling—like an aural painting. It's something you observe.

With video games, you are trying to create each individual sound event, which, when combined in an infinite number of ways, will be able to create another aural experience—something you partake in. One difference is that there's a lot more repetition in a video game. When you watch a film, the sound goes by in a linear fashion; you only hear those tracks at that given time within the movie.

In a video game, the player will hear sounds over and over. You need to be very conscientious when you create a sound or a melody that will be heard again and again that the end user won't get sick of it. In terms of sound quality, we treat games the same way we treat movies.

**Do you have some advice for those who are interested in developing their sound-design chops?**

The main thing I tell people when I give any lectures about sound design is to listen. Everybody has learned how to stop listening. [Instead, they] filter out those things around them that they find distracting. When I interview young sound designers, one of the things I ask them is to name five types of rain. Some answer, "Rain, I don't know: rain, more rain, heavy rain." I am looking for people who can hear and have an attention to detail—things like rain tapping on the window, rain hitting puddles, rain going down the gutter, the rhythm of the rain. Is it a slow *drip-drip* like a relaxed southern afternoon shower? Is it heavy pelting against a wood or plastic roof? All these sounds add to storytelling. You have to deal with sounds that evoke childhood emotions and that take you to a special place. I tell them to listen to the birds; listen to cars going by. Listen to anything that's machinery around your house. Listen to thunder and the way it echoes in the area that you live in, and how it slaps against the wall behind you. It's really stopping and relistening to everything around you. It's creating an aural vocabulary.

**So like your example of the Jacuzzi jets or the washing machine that sounds starshiplike, people should listen for sounds in their home that can be used as raw material for sound design?**

Yes. There is stuff that you can do in a home that would just blow your head away. Like taking [electric] razors and putting them in metal bowls and recording them. Or using the windows in your house on a windy day and opening them a little bit to make it sound like a wind storm or a hurricane or wind whispering across the wing of an aircraft. There are endless amounts of sounds that you can come up with.

**Do you generally recommend using stereo mics or stereo miking for capturing remote sounds?**

It really depends. It's the same thing as music, like saying when you mic a drum set or a guitar, what do you always use?

**Good point.**

I come from a music background, and I draw parallels between both industries. For example, when you're recording a gun, it's got a low thump that's kind of like a kick drum. So maybe I'll use an AKG D112 or an EV RE20 or a Sennheiser MD 421. I'll then use other mics to capture other frequencies that guns can make. Or maybe when I record glass breaking, which is kind of like percussion, I'll use a small-diaphragm condenser mic or a ribbon mic similar to overheads on a drum kit. I know it sounds corny, but I feel lucky to be able to be creative and work consistently on projects that I care about. I get to make noise for a living. EM

*Mike Levine is an EM senior editor.*

# Max Headroom

By Larry the O

## Gain staging in the digital domain.

Gain staging is the art of setting the multiple gain controls in any signal path to get the cleanest signal. Many of the rules of analog gain staging that pertain to balancing still apply to the digital realm, but when it comes to headroom and clipping, the rules of the game change completely.

A good place to begin in your pursuit of understanding how gain staging works in your DAW is to distinguish between two systems of calculation: floating point and fixed point. Most native DAWs use floating-point calculations, whereas PCM audio files and some DAWs, most notably Pro Tools, use fixed-point calculations.

### A Bit Too Much

Floating-point numbers use a fixed number of bits for the *mantissa*, or basic value; the remaining bits are used for the exponent. That amounts to a gain-ranging scheme controlled by the exponent; the mantissa is like a window of dynamic range that is moved up and down over the full dynamic range by changing the exponent.

Most commonly, 32-bit floating-point numbers with a 24-bit mantissa and an 8-bit exponent are used (exceptions include Cakewalk Sonar and Sonic Studio SoundBlade, which use 64-bit floating-point numbers). That gives the mantissa the same dynamic range as a standard 24-bit fixed-point PCM audio file. The gain ranging that is represented by the 8-bit exponent provides more than 1,500 dB of dynamic range. Plug-ins for native systems and add-on hardware like the Universal Audio UAD-1 PCI card also use 32-bit floating-point math.

In practice, it is almost impossible to clip an audio signal inside a 32-bit floating-point mixing engine—headroom isn't a problem. The problem arises when you try to send a signal with this huge dynamic range to an outside world that wants a 24-bit PCM signal—for example, when you bounce to disk or send a signal to the digital-to-audio converter (DAC) in your audio interface.

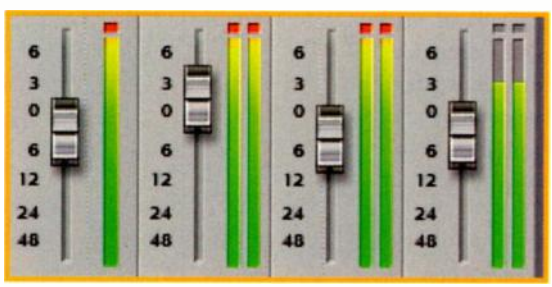



FIG. 1: It's 12 dB. Do you know where your clips are?

In recognition of this fact, the meters in most DAWs that have floating-point mixing engines actually reflect the dynamic range of the 24-bit PCM output. When the clip indicator lights up in your DAW, it doesn't mean the signal is clipping inside the DAW; it means that it would be clipped if it were output directly to a 24-bit fixed-point file or an audio interface. Avoiding clipping in a DAW mix is therefore usually as simple as inserting a master fader in your session and lowering it until its meter no longer shows clipping.

That being said, some attention to adding some gain staging earlier in the signal path may be useful. For instance, when you want to render individual tracks or subgroups to disk or directly to the output for external processing, it's convenient to already have them at an appropriate level. Furthermore, meters that are not pinned are more useful or at least more pleasing to look at. For one approach to gain staging in the digital domain, see "Step-by-Step Instructions" on p. 72.

### On the Other Hand

With Pro Tools and other fixed-point systems, the story is quite different. Pro Tools runs on Digidesign's proprietary TDM hardware, which has a 24-bit-wide, time-domain multiplexed bus. The program uses 48-bit double-precision, fixed-point math, so each signal inside the mixer uses up two TDM channels, or multiplexing time slots, to travel over the TDM bus.

Things get pretty complicated trying to figure out exactly what Pro Tools' meters are telling you, but Digidesign reserves a number of bits above 0 dBfs for headroom. As in native systems, channel clip indicators show that the signal will clip if it is sent to a DAC (as opposed to showing that it is clipping internally). On the other hand, the clip indicator of a master-fader strip does show actual clipping. It's possible, though not easy, to clip this output—a loud film mix with a lot of channels might be able to do so.

Pro Tools LE complicates the situation further because it is a native system and uses a 32-bit floating-point engine. The metering is complicated in LE, but two key meters—the prefader-channel meter and the final master-fader output meter—are

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## STEP-BY-STEP INSTRUCTIONS

**1**


Record signals to peak at no higher than –6 dB on the meters.

**2**

Start your mix with your faders set to an average position of half to two-thirds of the fader travel.

**3**

Use groups. Once you've gotten a good balance of elements within a group, using a group master won't mess with that balance.

**4**

Use master faders in your sessions. When a master-fader meter clips, simply lower the fader until the clipping stops.

**5**


Watch out for clipping in plug-ins. Lower the plug-in's input level if necessary.

**6**


If a track-level fader is inconveniently high or low, insert a gain plug-in to adjust the level.

24-bit fixed; therefore, *clipping* really means clipping.

To suit LE's floating-point engine, most RTAS plug-ins use 32-bit floating-point math. That makes the use of RTAS plug-ins in a TDM system a little messy, because the signals need to be converted between the two representations.

Gain staging has always been one of those skills that draw on both theory and practical experience. Although the details have changed with the migration to digital audio, that concept remains intact. Best results come from combining some understanding of what is happening under the hood with experimenting and listening on your own system. EM

*Larry the O recently finished floating-point scuba diving in Thailand.*

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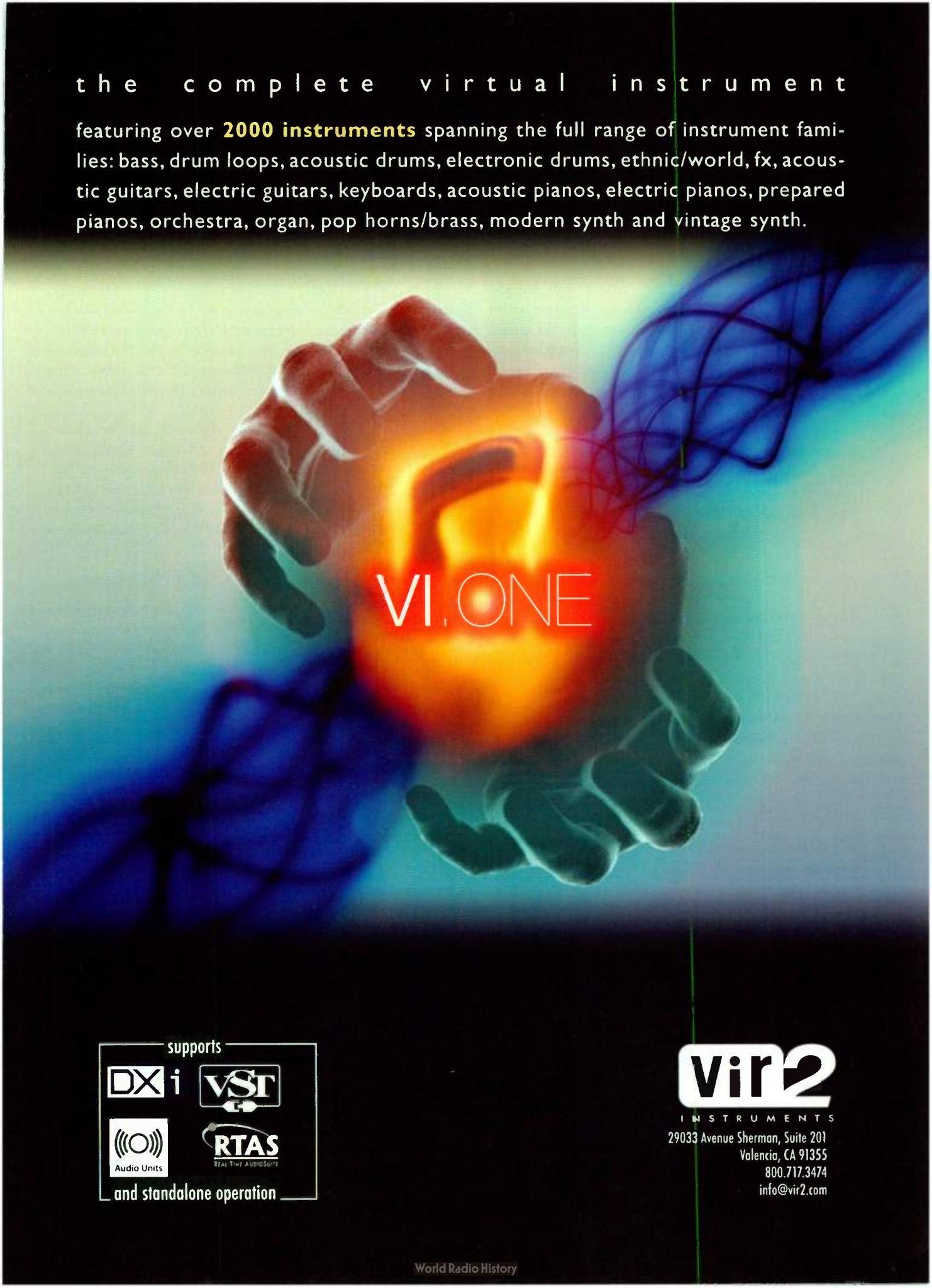


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# A Different Drummer

By Len Sasso

## A vocoder can turn your drum tracks upside down.

Using vocoders for nonvocal purposes is nothing new. Of the vocoder's two inputs, the carrier needs to have sustained notes with rich timbre. The speech input, which is used to shape the carrier, needs to be rhythmic and have quick changes in timbre. As an alternative to speech, the percussive and unpitched nature of drums makes them especially well suited to the task.

Full drum tracks, however, don't make the best vocoding sources; tailoring a drum part specifically for vocoding will produce more-interesting results. In this column, I'll describe how to build a Combinator based on Propellerhead Reason's BV512 Vocoder and Redrum modules to use as a general-purpose drum-vocoding effect (you'll find the Combinator and several Reason songs illustrating it in **Web Clip 1**).

### Combinatrix

Start by creating a new Combinator, which will automatically be cabled into your song's main mixer. Inside the Combinator create a Line Mixer, two Spider Audio modules (Spiders), a BV512 Vocoder, and a Redrum. Hold down the Shift key when creating these modules to prevent automatic cabling. Rename the two Spiders as Carrier and Speech.

Cable the Combinator's To Devices output to the Carrier Spider's Splitter input. Cable one of the Carrier Spider's Splitter outputs to channel 1 of the Line Mixer and cable another to the BV512's Carrier input.

Cable the Redrum output to the Speech Spider's Splitter input. Cable one of the Speech Spider's Splitter outputs to channel 2 of the Line Mixer. Cable both channels of another of its Splitter outputs to the left channel of its Merger input; that sums the Redrum outputs to a mono signal; cable the left Merger output to the BV512's Modulator input. Finally, cable the BV512's output to channel 3 of the Line Mixer and cable the output of the Line Mixer to the Combinator's From Devices input. I call this Combinator a Drumcoder.

I like to program the left two Drumcoder buttons to quickly solo the carrier and drums. Both buttons should solo channel 3 (the BV512) of the Line Mixer. The carrier-solo button should also toggle the BV512's Enabled switch between 1 and 0. The drums-solo button should also toggle the BV512's Dry/Wet control between 127 and 0 (see **Fig. 1**).

FIG. 1: The Redrum at the bottom provides the modulator input to the BV512 Vocoder.


### Care and Feeding

Reason sees the Drumcoder as an effects device. To use it, load a Reason module that produces a pad or ambient sound. It will automatically be cabled to the Drumcoder's Combi Input to become the carrier.

You could load a whole kit into Redrum, but you'll get better results choosing individual drum sounds based on their vocoding effect. Try standard drum sounds as well as sound effects. When you have some or all of the channels filled, move on to building patterns.

You can add interest to the Drumcoder in several ways. Insert a compressor with a high compression ratio between the Redrum and the BV512. Insert a long-tailed reverb in the Line Mixer's Aux bus and set the bus to pre-fader. Raising the carrier channel's send level will then add a hint of carrier without dominating the mix. Feed a pair of delay lines from the Redrum's Send outputs and use the Speech Spider to add their outputs to the BV512's Modulator input. Then, using different delay times, send some individual drum channels to each delay line.

I assign the Combinator knobs to the Line Mixer's Aux return, individual delay times, and the combined feedback of both delays. I assign a button to simultaneously enable both delays and another button to toggle the BV512 between 8- and 16-band operation. Consider using the LFO output of a Malstrom or Subtractor to modulate the Combinator knobs. EM

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*Len Sasso is an associate editor of EM. For an earful, visit his Web site at www.swiftkick.com.*

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# Preaching to the Converted

By Brian Smithers

## Modern A/D and D/A conversion demystified.

It seems we will never be rid of the analog-versus-digital debate. Far from a debate, really, it usually amounts to little more than hand-wringing over how perfect audio used to be before a horde of ones and zeros came along and ruined everything. You've heard it all before—no DAW will ever sound as good as tape; no virtual instrument will ever sound as fat as a real analog synth; and computers are eating our brains.

The reality for those who live in the here and now is that analog and digital audio are inexorably linked. The physical world is analog, so all audio must be analog by the time it reaches our ears. Virtually all audio distribution, however, is digital. The most critical step in audio production is getting the signal back and forth between the two domains. This article will explore the conversion of audio from analog to digital and back.

### Classical Conversion

The classic method of converting analog voltage to PCM digital audio is called *successive approximation*, which compares the input voltage with a series of progressively smaller reference voltages. Each time that the input exceeds the reference, a one is recorded; otherwise, a zero is recorded. Sixteen iterations are required to construct a 16-bit sample word. This is the same technique that you used in middle-school science class to weigh objects using a balance scale (see **Fig. 1**).

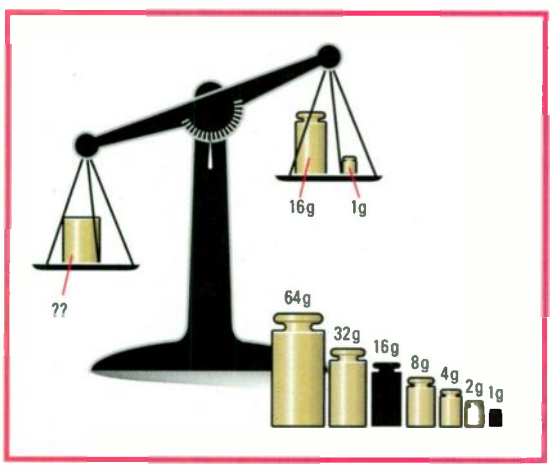


**FIG. 1: This graphic shows successive approximation as exemplified by a balance scale. An object of unknown weight is compared with a series of weights of decreasing value.**

Classic D/A conversion uses a ladder series of resistors of differing values. By switching them on in various combinations, the appropriate output voltage is achieved. Think of a series of lightbulbs of assorted intensities being switched on in different combinations to achieve varying levels of illumination, and you've got the idea.

These methods correspond directly to the PCM samples, with a stage of approximation or a value of resistor for each bit of the sample word. Various implementation challenges, however, have led developers to look for alternative conversion methods. One of the biggest challenges is the requirement for brickwall filters to prevent aliasing on input and imaging on output. Designing a sufficiently steep filter that does not distort within the audio band is difficult, to say the least. A second major challenge is achieving the required precision of the largest reference voltage in the A/D converter. The tiniest variation renders the smallest reference stage irrelevant.

### From Many, One

Modern A/D converters more often start by taking one-bit samples at a rate that is many times higher than the nominal sampling rate. The bitstream is then digitally converted to PCM data at the appropriate bit depth and sampling rate. Similar principles are applied to D/A conversion. In a perfect world, the difference between the two approaches would be negligible, but one-bit design is generally considered to give a more practical combination of quality and cost. The most common one-bit design is called a *sigma-delta* converter. In math-speak, *sigma* means *sum* and *delta* means *difference*, and a sigma-delta converter compares the input voltage with the sum of the previous input samples. If the input is larger than the reference, a one is recorded; otherwise, a zero is recorded. Because only one comparison is made for each word, rather than 16 or 24, there is no concern about how accurately the device can hold the input voltage through a series of comparisons. Additionally, the problem of precision relative to smaller approximation stages is not a concern when there is only one stage.

The simplicity of a sigma-delta converter allows it to run at much higher sampling rates. Capturing more samples than are required by the nominal sampling rate is called *oversampling*. Less information is contained in each sample, but there are many more samples. If multibit samples are like film, in which each frame

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contains a detailed picture and a modest number of frames are displayed per second, then sigma-delta conversion is like laser animation, in which the position of a single point of light is updated so quickly that our eyes perceive solid lines and complex graphics.

Oversampling raises the Nyquist frequency, thereby relaxing the requirements of the input filter. If sampling takes place at 64 times the nominal sampling rate of 44.1 kHz, frequencies as high as 1.4 MHz can be captured without aliasing. A moderate and extremely well-behaved analog lowpass filter is sufficient to prevent foldover.

Although quantization error is quite large in a one-bit system, oversampling reduces quantization noise within the audio band. Quantization noise is always spread uniformly from 0 Hz to the half-sampling (Nyquist) frequency. By raising the half-sampling frequency, oversampling moves most of the noise beyond the range of our ears (see **Fig. 2**; note that the sampling rate is referred to as *Fs*, and the half-sampling rate is *Fs/2*).

In order to reduce the quantization noise further, *noise shaping* is applied. As the captured samples are fed back to be compared with the input voltage, low-level noise called *dither* is applied to randomize its pattern. The feedback loop of dithered quantization noise being added to the input signal increases the total noise, but it shifts the spectrum of the noise so that it is more prominent at the Nyquist frequency and reduced in the audio band, as shown in Fig. 2.

Once the one-bit data has been captured, a *decimation filter* converts it to a standard PCM representation. At the same time, it lowpass-filters frequencies above the Nyquist frequency of the output sampling rate. This also removes the ultrasonic quantization noise. Although this is still a very steep filter, it is easier to implement in the digital domain, alleviating the distortions within the audio band characteristic of an analog brickwall input filter.

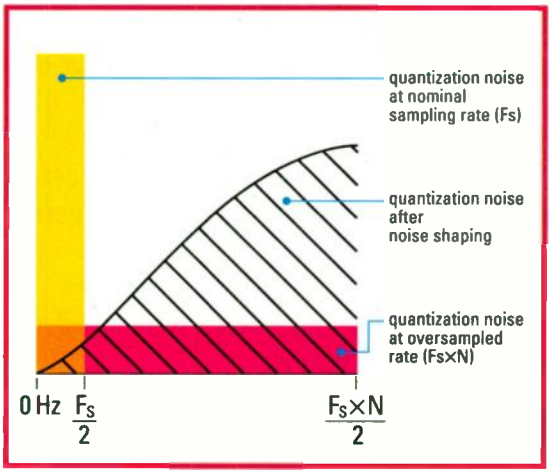


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**FIG. 2: Quantization noise is distributed evenly from 0 Hz to the Nyquist frequency. Oversampling A/D conversion raises the Nyquist frequency, reducing the level of quantization noise in the audio band. Noise shaping tilts the distribution further into the ultrasonic band.**

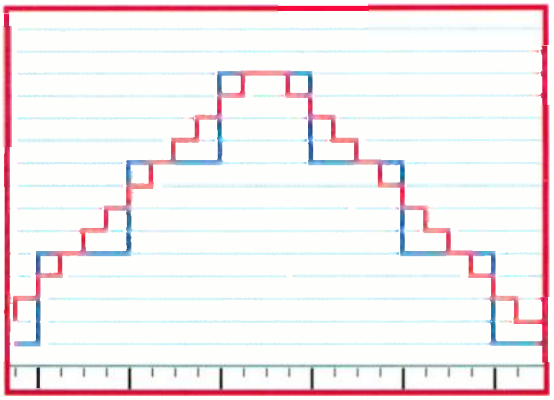

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**FIG. 3: The stair-step effect is the squaring off of the output waveform, inducing undesired high-frequency components. Oversampling D/A conversion creates smaller stair steps, effectively shifting the undesired frequencies into a range where they are more easily filtered.**

## Faster, Faster

The steep output filter found in all D/A converters is also known as an *anti-imaging filter*. During each sample period, the converter emits a constant voltage that results in a squared-off wave (see **Fig. 3**). This is known as the *stair-step effect*. Just as the corners of a square wave when viewed on an oscilloscope represent the wave's overtones, the stair-step edges of the output wave constitute undesirable multiples of the actual signal. These images of the signal occur near integer multiples of the sample frequency and extend as low as the Nyquist frequency. The output filter is sometimes called a *smoothing filter* because it removes these corner frequencies and leaves only the continuous waveform of the intended signal.

Like the input filter, the output filter must be extraordinarily steep, creating severe design problems in the analog domain. Oversampling D/A converters raises the effective sampling rate by interpolating new samples between the existing samples. That pushes the image spectra high enough that lower-order (less steep) filters can adequately remove them without distorting the audio band.

As with the successive approximation A/D converter, a classical ladder D/A converter requires extreme precision from the largest-value resistor to keep from nullifying the action of the lowest-value resistor. One-bit D/A conversion alleviates this worry by requiring a single capacitor that is charged for a one and discharged for a zero. The multibit PCM stream is converted to a one-bit stream that is then filtered to produce the output signal.

The relatively simple design of one-bit oversampling A/D and D/A converters makes them attractive for reasons of cost and quality. Indeed, the shortest path between analog and digital is often a single bit. EM

*Brian Smithers is Course Director of Audio Workstations at Full Sail Real World Education in Winter Park, Florida. His latest book is* Mixing in Pro Tools: Skill Pack *(Thomson Learning, 2006).*

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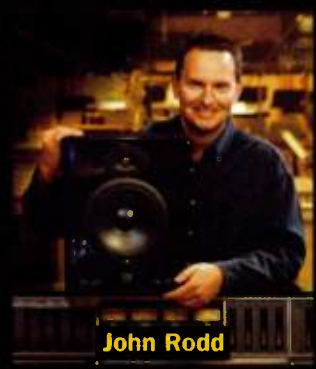

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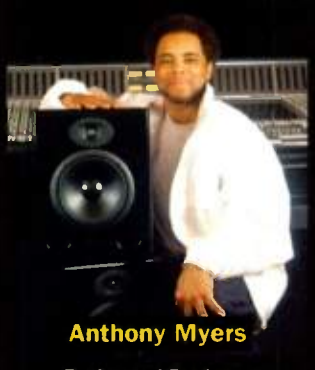

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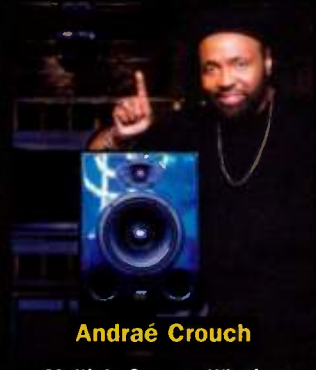

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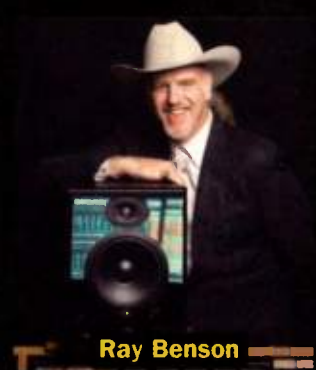

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Right. Whenever you're looking at any contracts that have to do with your copyrights, you want to be extremely careful because that's the core of your whole business: copyright—your intellectual property.

**Let's talk about the process of copyrighting a song or composition. I gather that it's not as simple as some people think.**

Musicians and songwriters sometimes really mess up the copyright thing. Because even though it's a short form, you have to figure out what the form is that you're using. You have to figure out whether there are going to be coauthors on there, who owns the copyrights, whether it's going to be a work made for hire, and whether it can even be a work made for hire legally. It's really important that songwriters and composers register their copyrights, or artists and producers their sound recordings—whatever—with the U.S. Copyright Office at the Library of Congress and not with some online scheme. Anybody (except, of course, a lawyer) who tells you that they can register your copyright for you is scamming you.

**You can save money by copyrighting a group of songs together. But is there anything to watch out for in that instance?**

The mistake that people make when they copyright a collection of works themselves is that they don't list all the titles. And when you search the Copyright Office records or are trying to prove that you have a registered copyright in order to enforce your rights in court, you won't find that title registered. If a bunch of songs are registered under a single title, it could cause huge legal problems. There's a form called Form CON [a *continuation sheet*] that is available online just like all the other copyright forms [go to www.copyright.gov/forms]. On that form you can list the additional titles, and they'll be covered. Be sure to do that; so many people just give the whole collection one name like "My Song Collection, v. 1." Great. What was in there? And it becomes an evidentiary issue of fact. And it can really sideline your lawsuit if you're trying to enforce your copyright. Also, the Copyright Office has just announced plans for rate changes that are scheduled to go into effect sometime between October 1, 2007, and January 1, 2008, that will affect the cost of registering additional titles in collections. It will cost an extra $1 for each title on an electronic filing (electronic filing is supposed to start around July 1, 2007) and $3 more for each title on a paper application. Basic electronic filing fees are supposed to be $35, while paper filing will remain $45.

**Let's say that your music has been properly copyrighted, and a producer offers to produce your music at some point. You still need to work out an agreement to protect yourself, right?**

Yes, because the producer may actually be a co-owner of your copyrights with you. If you don't want that to be the case, you've got to have a written agreement in which he or she gives that up but gets other certain rights.

**The producer gets that copyright—that is, the co-ownership of the sound recording—by default?**

Possibly—as an author of the sound recording. Because if somebody is actively creating, not just running the machine, they could be entitled to co-ownership of the copyright. Engineers probably wouldn't be. But who draws the definitive line between an engineer and a producer? It's certainly a gray area that needs to be clarified through a written agreement.

**Clearly there are issues regarding the copyright of sound recordings. But what about the copyright of compositions?**

With the composition you can get into problems when a songwriter brings the song into the band, and then, say, the drummer does a little drum thing, the rhythm section guys do a little arrangement, and the piano guy does his arrangement. Are they then coauthors of the song? It's actually a matter of intent. According to copyright law, their contribution doesn't have to be substantial; it just has to be more than nominal. Songwriters also have to be careful about producers claiming some of their song copyrights, as in when the producer helps the artist work out the songs in the studio, unless, of course, their intention is to share the song copyrights. It could be the producer or the singer who says, "Hey, it would sound better if I sang these words differently. Let's change them around." Something important about copyright law is that unless there's a written agreement, coauthors share equally [rather than] in proportion to their contribution. Without an agreement, you could write all the music and most of the words, and somebody else, whose only contribution was helping with the words, could own 50 percent of the words and music of the total song. EM

*Mike Levine is an EM senior editor.*

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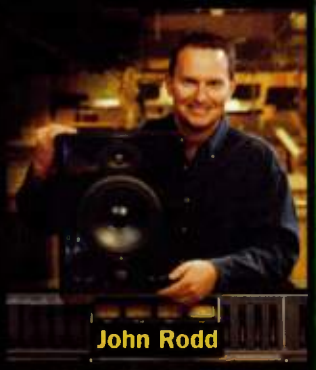
Multi-Platinum Grammy-Winning
Engineer / Producer



**John Rodd**

Orchestral Scoring Recordist
Music Scoring Mixer

**"Prince Charles" Alexander**

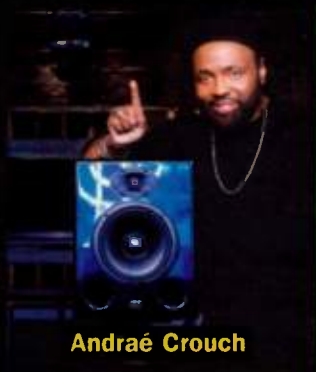
Multi-Platinum Grammy-Winning
Engineer / Producer

**Anthony Myers**

Engineer / Producer
Sound Designer

**Gerhard Joost**

Multi-Platinum Engineer
Mixer / Producer



**Andraé Crouch**

Multiple Grammy-Winning
Gospel Singer / Producer / Pastor

**Lionel Jarvis**

Multi-Platinum Musician
Musical Director

**DJ Ron G**

Remixer / DJ

**Steve Pageot**

Grammy-Winning Producer
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# Q&A: Bev Green

By Mike Levine

## Part 2: Legal advice for composers and songwriters.

This is the second of two columns featuring music attorney Bev Green, a veteran entertainment lawyer in the San Francisco area who practices at the firm of Green and Green (www.musiclawyer.com). Besides representing a who's who of musical talent, Green has negotiated licensing deals involving artists such as Destiny's Child, Will Smith, and Alan Jackson. She has also been active in licensing material for new-media projects such as Hit Clips, the micro music player for kids from Tiger Electronics/Hasbro, and various musical products from Fisher-Price.

In part 1 of my interview with Green (see "Music Business Insider: Q&A: Bev Green" in the May 2007 issue of EM), she offered some general legal advice for musicians. This time, she focuses on the legal issues that songwriters and composers face.

**Like bands and artists, songwriters and composers need to be careful about signing agreements, correct?**

Yes. Generally, I would say don't give up your music copyrights (called the *publishing*)—at least not without very careful advice and contract negotiations.

> "Musicians and songwriters sometimes really mess up the copyright thing."

**Is that the writer's share of the publishing that you're referring to?**

No, the publishing is the ownership and control of all of the song. It's a little bit confusing because the person who writes the song—that's the author—is the copyright owner to begin with. The copyright law was specifically written to protect writers so they can't transfer their ownership, or publishing, without a written agreement signed by them. Otherwise, it would be claimed to be happening all the time. The thing is that in that written agreement, a writer could actually give up the copyright to their song and not have in that contract a contractual right to a writer royalty, the so-called writer's share. BMI and ASCAP arbitrarily divide the royalties into a writer's share and a publisher's share. If you're the writer-publisher, you keep the whole thing. But even if you've signed over all of your publishing and you didn't get any writer royalty, you can still get your public performance fees from BMI or ASCAP (if you join one of them), because they insist on paying the writer directly, to save you from totally screwing yourself over.

**Would this count in a broadcast situation? If you wrote music for, say, a TV network and signed away your writer's share, would you still be eligible for payments from ASCAP or BMI?**

Actually, I've seen that situation, because I do represent a lot of people who write for TV and film. If you're writing original music for a show, the production company will usually want a contract with you where they acquire all the publishing [rights] in the music. But they will usually let you keep the writer's share of at least the performing rights, which you would get some of from television performance but not from motion picture or theater performance—at least not in the U.S.—because of the way the law has developed.

**If you do a composition that's a "work made for hire," do you give up a lot of rights?**

Yes, you give up entirely the ownership of the copyright from the inception of the creation of the work. In other words, the employer becomes the legal author. You also give up your possible future rights to terminate an assignment and even give up the right to have your name listed on the copyright registration as the legal author. Any rights you may want, such as credits or royalties, you will have to negotiate for and include in the written contract. Not all types of work can be considered as work made for hire. It's a complex subject, but suffice it to say that for a musical work specially commissioned for use in a motion picture or TV show to be considered as a work made for hire, there would need to be a signed contract stating that that's the case (unless the work is done in a true employment situation).

# "We Had a Hit Single with Jesse McCartney, and it all Began with TAXI"

*Andy Dodd and Adam Watts – TAXI members*
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Adam and Andy's success through TAXI is a little bit different from all the other stories you've probably heard. They got their *biggest* deal after their membership ran out!

Here's how it happened: "We joined TAXI in 2001 and found that it was a great motivator for us. We were members for two years. We learned a lot, wrote a ton of songs, and got a few film and TV placements -- some through TAXI, and some on our own.

We submitted a song we wrote with Jenn Shepard called "You Make Me Feel" to one of TAXI's Industry Listings. We didn't hear anything back for a while and eventually our TAXI membership ran out. Thankfully, we began to get so busy with production and writing gigs that we decided to wait and renew our membership at a later date.

Little did we know that TAXI had sent our song to a production/management company that was looking for material for a young, male Pop artist they were developing.

Later that year, Jesse McCartney's managers called us saying they had just heard "You Make Me Feel" on a CD they got from TAXI and wanted to have him cut the song. Although Jesse decided not to record "You Make Me Feel", his managers asked us to write more songs for him. We wrote a handful and they ended up putting his vocal on two of the tracks we produced, "Take Your Sweet Time" and "Beautiful Soul".

"Beautiful Soul" got played on Radio Disney, and Jesse's management got the song to a label executive at Disney. Soon after, Jesse was signed to Hollywood Records. "Beautiful Soul" became his first single, and we both signed publishing deals with Disney Music Publishing.

Jesse McCartney's album (entitled "Beautiful Soul") has gone Platinum in the U.S. and Australia.

"Beautiful Soul" went to #3 on Radio and Records CHR Pop Chart, #5 on Billboard's Top 40 Chart, #19 on Billboard's Adult Top 40 chart, it's a Platinum Digital Single Download, it's on the Gold-selling 'Cinderella Story' Motion Picture Soundtrack, the Gold-selling 'That's So Raven' TV Soundtrack, and the video was nominated for Best Pop Video at a 2005 MTV Video Music Awards."

All of this came about because Adam and Andy sent a song to TAXI. Call for our free information kit!


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**Would you still be entitled to ASCAP or BMI monies?**

Yes. Even though the publisher is considered to be the author for copyright purposes, they are legally required to identify the actual author when they put in their registrations with ASCAP and BMI, and the composer would get paid directly.

**So if a production company shoves a contract in your face, you would recommend showing it to an attorney first—even if you don't have one yet?**

Right. Whenever you're looking at any contracts that have to do with your copyrights, you want to be extremely careful because that's the core of your whole business: copyright—your intellectual property.

**Let's talk about the process of copyrighting a song or composition. I gather that it's not as simple as some people think.**

Musicians and songwriters sometimes really mess up the copyright thing. Because even though it's a short form, you have to figure out what the form is that you're using. You have to figure out whether there are going to be coauthors on there, who owns the copyrights, whether it's going to be a work made for hire, and whether it can even be a work made for hire legally. It's really important that songwriters and composers register their copyrights, or artists and producers their sound recordings—whatever—with the U.S. Copyright Office at the Library of Congress and not with some online scheme. Anybody (except, of course, a lawyer) who tells you that they can register your copyright for you is scamming you.

**You can save money by copyrighting a group of songs together. But is there anything to watch out for in that instance?**

The mistake that people make when they copyright a collection of works themselves is that they don't list all the titles. And when you search the Copyright Office records or are trying to prove that you have a registered copyright in order to enforce your rights in court, you won't find that title registered. If a bunch of songs are registered under a single title, it could cause huge legal problems. There's a form called Form CON [a *continuation sheet*] that is available online just like all the other copyright forms [go to www.copyright.gov/forms]. On that form you can list the additional titles, and they'll be covered. Be sure to do that; so many people just give the whole collection one name like "My Song Collection, v. 1." Great. What was in there? And it becomes an evidentiary issue of fact. And it can really sideline your lawsuit if you're trying to enforce your copyright. Also, the Copyright Office has just announced plans for rate changes that are scheduled to go into effect sometime between October 1, 2007, and January 1, 2008, that will affect the cost of registering additional titles in collections. It will cost an extra $1 for each title on an electronic filing (electronic filing is supposed to start around July 1, 2007) and $3 more for each title on a paper application. Basic electronic filing fees are supposed to be $35, while paper filing will remain $45.

**Let's say that your music has been properly copyrighted, and a producer offers to produce your music at some point. You still need to work out an agreement to protect yourself, right?**

Yes, because the producer may actually be a co-owner of your copyrights with you. If you don't want that to be the case, you've got to have a written agreement in which he or she gives that up but gets other certain rights.

**The producer gets that copyright—that is, the co-ownership of the sound recording—by default?**

Possibly—as an author of the sound recording. Because if somebody is actively creating, not just running the machine, they could be entitled to co-ownership of the copyright. Engineers probably wouldn't be. But who draws the definitive line between an engineer and a producer? It's certainly a gray area that needs to be clarified through a written agreement.

**Clearly there are issues regarding the copyright of sound recordings. But what about the copyright of compositions?**

With the composition you can get into problems when a songwriter brings the song into the band, and then, say, the drummer does a little drum thing, the rhythm section guys do a little arrangement, and the piano guy does his arrangement. Are they then coauthors of the song? It's actually a matter of intent. According to copyright law, their contribution doesn't have to be substantial; it just has to be more than nominal. Songwriters also have to be careful about producers claiming some of their song copyrights, as in when the producer helps the artist work out the songs in the studio, unless, of course, their intention is to share the song copyrights. It could be the producer or the singer who says, "Hey, it would sound better if I sang these words differently. Let's change them around." Something important about copyright law is that unless there's a written agreement, coauthors share equally [rather than] in proportion to their contribution. Without an agreement, you could write all the music and most of the words, and somebody else, whose only contribution was helping with the words, could own 50 percent of the words and music of the total song. EM

*Mike Levine is an EM senior editor.*

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# ADAM PROFESSIONAL AUDIO A7

## Big sound in a little package.

By Eli Crews

Over the past few years, I've heard a lot about ADAM studio monitors, but they seemed a little beyond my price range. With the A7 model, ADAM has introduced its first self-powered monitors for under $1,000 a pair, putting them well within reach of most project studios—exactly what a lot of engineers have been waiting for.

The A7 has two integrated amplifiers—one for the tweeter and one for the woofer—each supplying 80W of RMS power. The crossover frequency is 2.2 kHz.

FIG. 1: The A7's diminutive size belies its large sound and great depth of field.

The A7 is similar in size to other close-field monitors, but it stands out in a crowd visually (see Fig. 1). The top front corners are beveled to reduce surface reflections from the tweeter, giving the monitors a looming, sci-fi look. Instead of a dome tweeter, the A7 has a small, slatted grille with what looks like yellowed air-filter material nesting behind it. This is the unique Accelerated Ribbon Technology (ART) tweeter, which, according to a thorough but somewhat opaque explanation on ADAM's Web site, is four times as efficient at moving air than the average metal dome tweeter. That may help explain how the A7 can reproduce frequencies up to 35 kHz (±3 dB).

The A7's 6.5-inch woofer is striking as well, with a Rohacell/Carbonfibre cone that has the appearance of black tweed. Rounding out the front of the monitor are a power switch, an input gain control, and a round bass-reflex port.

The input gain control on the review units is dome shaped, with a little oval dimple indicating where the knob is set. The control has a range from –∞ to +6 dB. Although aesthetically pleasing, this control isn't very precise—it's tough to figure out where the oval should lie relative to the gain markings surrounding the pot, making it difficult to get the same setting on a pair of monitors. However, ADAM says that the newer units have a more utilitarian cylindrical knob.

### GUIDE TO EM METERS

5 = **Amazing**; as good as it gets with current technology
4 = Clearly above average; very desirable
3 = Good; meets expectations
2 = Somewhat disappointing but usable
1 = Unacceptably flawed

**Specifications tables for EM reviews can be found at www.emusician.com/specs.**

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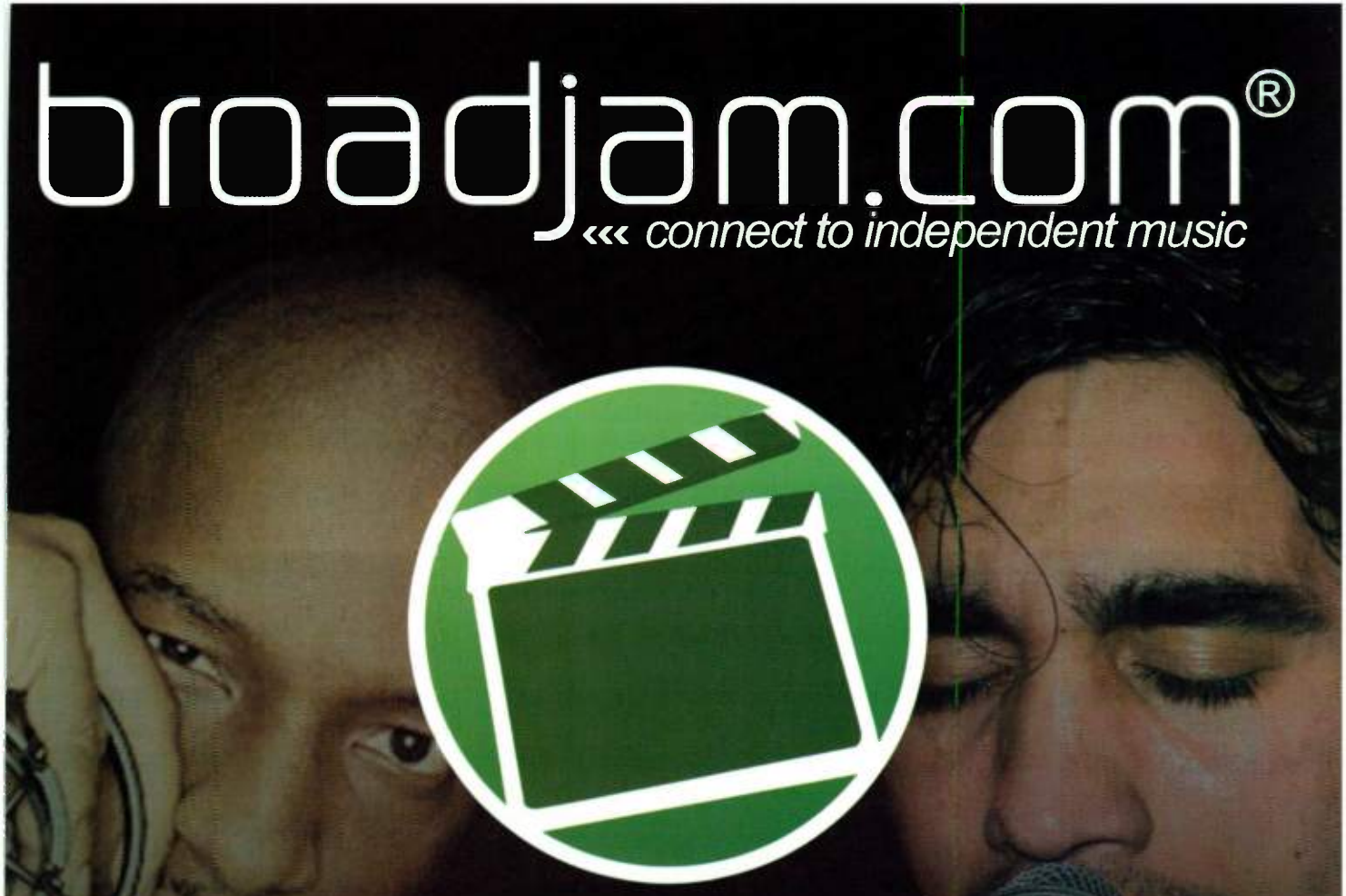

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On the rear of the monitor are an XLR and RCA audio input, a standard IEC power cable receptacle, and a hefty heat sink. Small recessed pots let you tune the speaker to your control room. The low shelf has a corner frequency of 150 Hz, the high shelf is set at 6 kHz, and each knob gives you ±6 dB of active EQ. There is also a control for the tweeter level, which lets you cut or boost its amplifier's gain by 4 dB. ADAM recommends using this as a last resort only, because it matches the woofer and tweeter levels in its Berlin factory. (Some of the A7's parts are manufactured in Singapore.) Nonetheless, the tweeter control is there if you need more or less high end than the high-shelf control can accommodate.

The EQ and tweeter gain controls themselves are a tad flimsy. Again, it's hard to tell exactly where the arrow (which is adjustable with a small flat-head screwdriver) points, because the pots are a little crooked in their housing and the detents are too close together.

## As Heard on TV

In my studio I compared a pair of A7s with my Bag End M6s, and right away I noticed the A7's clarity, depth, and tight bottom end that I've heard people raving about. (I also have a pair of Genelec 1030As in my studio, but I removed them in order to put the A7s in my control room's sweet spot.) Next to the M6s, the A7s had slightly more treble and a little less energy in the high-mid region, which served as a nice complement when comparing mixes between the two.

To hear the A7s in a different room, I took them to Philo TV, a postproduction house in San Francisco where I also work. With the help of engineers Christian Hanlon and George Sakellariou (who uses a different brand of ribbon-tweetered monitors), I A/B'd them against the studio's set of 1030As.

The three of us compared the two sets of monitors using tones, pink noise, and our favorite records, and we spent a couple of weeks using the A7s to mix projects destined for television. Because



PRODUCT SUMMARY

**ADAM PROFESSIONAL AUDIO A7**

powered monitor
$499.50 each

| | |
|---|---|
| AUDIO QUALITY | 4 |
| VALUE | 4 |

RATING PRODUCTS FROM 1 TO 5

**PROS:** Clear, open sound. Extended frequency response, especially for a small speaker. Easy to monitor on for long periods of time.

**CONS:** Controls are imprecise.

MANUFACTURER
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the commercials we worked on had a good deal of dialog, this experience gave me a very good sense of how the A7s work for mixing both music and voice.

> The A7 monitors, frankly, made the added treble more listenable.

One of our first conclusions was that the A7 has a deeper soundstage than the 1030A. The 1030A seemed to add extra compression to the material, while the A7 let the dynamic range breathe in a way that extended the three-dimensional imaging much more. The A7 pair also gave me a very strong sense of the center image, even though they were physically farther apart in the studio than the 1030As. It felt like the sweet spot, where everything comes together phasewise between the two monitors, was nice and wide on the A7s.

Our main impression was that the A7 doesn't sound as if it has nearly as much high-end reproduction as the 1030A. However, the benefit is that the A7's high end sounded smoother, which made long periods of listening to them more pleasurable. Because we were mixing for TV, it was essential to have the mixes turn out bright, and the A7s, frankly, made the added treble more listenable.

But the name of the game when it comes to mixing is translation—how good your mix sounds on a variety of real-world playback systems, such as TV speakers, home-stereo systems, earbuds, computer speakers, and car stereos. The mixes I did on the A7s passed the car test, the MP3 test, and the home-stereo test remarkably well.

## Up and ADAM

Choosing a pair of studio monitors is a very personal decision that can be made only by auditioning all of the available flavors. That said, you would be hard-pressed to find a better speaker at this price point. (The A7 also comes with a two-year warranty.)

If you're looking for a great-sounding monitor that lets you dig deep into your mixes, give the A7 a listen. You just might like what you hear.

*Eli Crews records at his studio, New, Improved Recording (www.newimprovedrecording.com), in Oakland, California.*

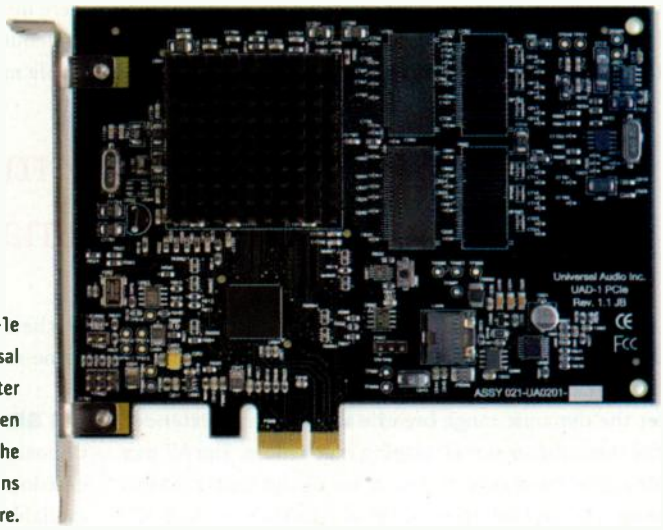


FIG. 1: You'll need the UAD-1e expansion card to host Universal Audio plug-ins in a computer with PCI Express slots. When bundled with the Expert Pak, the card comes with five plug-ins and a $750 voucher to buy more.

UNIVERSAL AUDIO

UAD-1e Expert Pak (Mac/Win)

# Take a load off your CPU with some outstanding plug-ins.

By Larry the O

Universal Audio (UA) was established to further the vision of recording pioneer Bill Putnam, whose company was the first to bear the name. Since Putnam's sons Bill Jr. and James resurrected UA in 1999 and later merged it with Kind of Loud Technologies, the company has taken a two-pronged approach. It has meticulously re-created analog hardware, including a few of Putnam's classic inventions. It has also developed a family of native audio plug-ins that includes emulations of processor hardware. Most recently, the company has been popping out models of analog gear from Roland and Neve.

UA's plug-ins run exclusively on its DSP expansion cards, and they are the only software that does; no third-party plug-ins are available for either the original UAD-1 acceleration card or the newer PCIe-bus version, the UAD-1e (see Fig. 1). Fortunately, this exclusivity is fine, because the plug-ins sound extremely good. And even though they require a UAD card to run, they appear in your audio software like any other plug-ins.

The plug-ins that accompany the UAD-1e in UA's Expert Pak bundle—the CS-1 channel strip, 1176SE limiting amplifier, Pultec EQP-1A program equalizer, Nigel guitar processor, and RealVerb Pro reverb—are the same five that originally shipped with the UAD-1. Although the UAD-1 and UAD-1e work in different buses, they have the same custom DSP chip.

The UAD-1e is supported on both Mac OS X Universal Binary and Windows 2000/2003/XP systems and works with native hosts only. In Digidesign Pro Tools, the UAD-1e supports RTAS using a free version of FXpansion VST to RTAS Adapter that UA provides, but it doesn't support TDM. The plug-ins run at sampling rates from 44.1 kHz to 96 kHz, and all but a few of them also run at 176.4 and 192 kHz. A system can contain as many as four UAD-1e cards, and a PC that has both PCI and PCIe slots can mix the UAD-1 and UAD-1e.

## Inward and Onward

Installing the card and its drivers into my quad-core 3 GHz Mac Pro was simple and seamless, and the card never again required my attention. Installing the software (version 4.5) took an extra step to authorize the

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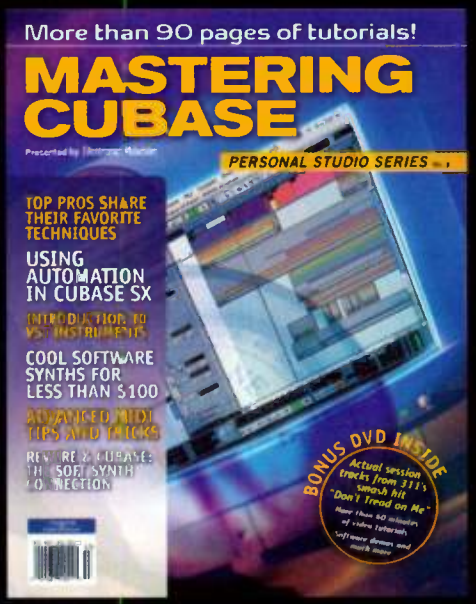


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plug-ins, but that process was also painless.

The UAD-1e is available in either the Express Pak ($599) or the Expert Pak ($1,299). Both packages include the UAD-1e card, the five plug-ins I mentioned, and a voucher for buying additional plug-ins from UA's Web site. The Express Pak includes a $100 voucher, and the Expert Pak includes a $750 voucher. If you assume that the card costs $499, the Expert Pak doesn't work out to be a greater bargain unless you calculate using street prices, but I promise you'll never regret having the extra plug-ins.

FIG. 3: Neve 33609 is a versatile and powerful compressor/limiter. It has a sweet sound with a subtler coloration than some of UA's other vintage compressors.

At the January 2007 NAMM show, UA announced two more UAD-1e developments. The UAD-1e Extreme Pak ($2,499) includes all of UA's plug-ins—every last one of them. The UAD-Xpander, a new ExpressCard for laptops, is available in three bundles with the same plug-ins as the UAD-1e: the UAD-Xpander Xtreme Pak ($2,599), the UAD-Xpander Xpert Pak ($1,699), and the UAD-Xpander Xpress Pak ($1,199).

The UAD-1e's excellent PDF owner's manual contains a tremendous amount of useful knowledge, from providing general information about signal processing and how to use the UAD-1e to revealing the idiosyncrasies of the original hardware (which may be essential to understanding the original's appeal) and explaining the plug-ins' added features. While you can apply nearly all the plug-ins to good result just by instantiating and playing with them, you simply cannot get the most out of them without reading the manual.

## Applied Theory

I'll begin by discussing the UAD-1e's five included plug-ins, and I'll continue with a rundown of additional plug-ins you can buy. I also refer you to the November 2002 review of the original UAD-1, as well as the March 2006 review of the Plate 140 reverb plug-in (both are available online at www.emusician.com).

FIG. 2: CS-1 packs four processors into a single plug-in that provides EQ, compression, delay, and room simulation.

CS-1 is a collection of four processors in one plug-in: the DM-1 delay modulator, RS-1 room simulator, and EX-1 compressor and 5-band parametric equalizer (see Fig. 2). If you don't need them all, you can instantiate them separately to reduce DSP demand. CS-1's utility is great and its sound is excellent, though not quite as good as some of UA's plug-ins dedicated to only one function. The compressor is nice, but not quite up to the 1176 models.

I use the 1176 plug-ins a whole lot—mostly 1176LN, but also the SE version. As UA says, the sonic difference between them is slight and not really noticeable unless the source is very exposed. I've had the opportunity to use the original UREI 1176 hardware, and the model really captures its distinctive but broadly applicable sound.

EQP-1A is for shaping, not surgery. The original Pultec EQP-1A was known for its beautiful, round warmth at high levels, and the plug-in represents that behavior well. I put a slightly wimpy kick drum in, cranked it up at 80 Hz, and got back a fat, punchy kick drum, matching my first experience with Pultec hardware.

RealVerb Pro worked well on delicate sources like vibraphone and vocals, as well as on drums and guitars. Whereas most digital reverbs let you adjust acoustic attributes such as diffusion, RealVerb lets you adjust architectural attributes such as wall materials. The cool thing about RealVerb, though, is that it has some acoustic parameters too, so you get the best of both worlds. Universal's DreamVerb plug-in, which is essentially the next iteration of the RealVerb engine, provides even more-intricate control of architectural considerations.

Nigel is a guitar suite that comprises five modules: the Preflex amplifier/cabinet simulator, Gate/Compressor, Phasor, Mod Filter, and Tremolo/Modulated Delay/Echo. It's kind of like a collection of stompboxes with an amp modeler. You can instantiate each module independently, and I found myself using the modules separately more than together.

Through the graces of Universal Audio, I was able to try a variety of Powered Plug-Ins beyond those included with the Expert Pak. UA has a lot of plug-ins in its catalog, and nearly all of them are outstanding. My particular favorites are the dynamics processors, including 1176LN and 1176SE, Fairchild 670, LA-2A, Neve 33609, and the highly transparent Precision Limiter. Plate 140 (a model of the EMT 140 plate reverb) is the first reverb emulator I have heard that sounded anywhere close to a real plate. The EMT 140 has a bright, explosive attack and very high density that no digital plate before Plate 140 has managed to reproduce.

## All Things Equal

Universal Audio makes 21 plug-ins for the UAD-1e card. You can also get two utilities (Track Advance and Delay Compensation), individual versions of all the modules in CS-1 and Nigel, and reduced-DSP versions of the 1176 compressor and the Neve models. That's a fair amount of software. Although I don't have room to adequately report on everything, I had the opportunity to try nearly all of them. Individual plug-ins cost between $79 and $249, and UA also offers bundles containing two or three related plug-ins.

In addition to Pultec EQP-1A and CS-1's 5-band parametric EQ, UA offers several other equalizers. Cambridge EQ ($149) is modeled after Sony's Oxford EQ—something I can say but UA can't. Others include Neve 1073 ($249), Neve 1081 ($249), and Pultec Pro ($79), which adds a midrange band to Pultec EQP-1A. Another EQ called Precision Equalizer ($199) is designed for mastering. They all sounded terrific to me, but I used the Cambridge and Pultec EQs the most.

Cambridge EQ's five parametric bands and high- and low-cut filters sport numerous features beyond the usual frequency, Q, and boost and cut parameters. For example, four different topologies are available for the cut filters: coincident pole, Bessel, Butterworth, and elliptical. Each topology has its own characteristics and offers several choices of filter order. Three parametric types, which apply globally


FIG. 4: Unlike some plug-ins designed to look exactly like the hardware they're emulating, RealVerb Pro takes advantage of computer graphics to make every user parameter clear and easy to edit.

to all parametric bands, use the same algorithm but have different behavior for gain versus Q. In addition, each parametric band can be individually switched to shelving or peaking.

Neve 1073 emulates one of the most famous of all audio equalizers. With two shelving bands, one semi-parametric band (Q is not user controllable), and a low-cut filter, it is not the most surgical of EQs, but it has that amazing Neve sound. The virtual front panel is nearly all rotary pots, including two concentric pots. Although this layout makes readability quite poor, vertical dragging is enabled and saves the interface from being hard to operate. It also helps that clicking repeatedly on the band-response symbol above any of the EQ pots cycles through that band's available frequency values. Neve 1081 is another Neve classic; it has four bands and high- and low-cut filters.

## Weightless Compression

Compressors include the aforementioned 1176, LA-2A ($149), Fairchild 670 ($149), Neve 33609 ($249), Precision Multiband ($249), and Precision Limiter ($199). Favorites are really hard to call in this area, because each has a distinctive sound. I compared Precision Limiter with several other limiter plug-ins, including Waves L1 and L2, and found Precision Limiter to be the most transparent, even when adding a fair amount of oomph.

Neve 33609 was a very pleasant surprise (see **Fig. 3**). I have never had the good fortune to encounter a real 33609 in the flesh, so I was completely unfamiliar with it before trying UA's plug-in version. In spite of being quite costly in terms of DSP, it immediately became a favorite. It's difficult to describe the sound; it's not squishy like the Fairchild 670, and it's not transparent like the Precision Limiter. It has a wonderful sound quality that is not too heavy handed. Each channel of the 33609 has a compressor feeding a limiter, and the two channels can be linked in stereo (a feature lacking on the original unit) or run separately.

It is safe to say that a Fairchild 670 is something very, very few of us will ever actually own. Given the cost, size, weight, and maintenance, it is questionable whether one would even want to—that is, considering that the UA plug-in makes the same sound available for a lot less cost. The 670's controls have considerably more subtlety than might appear at first glance, but it offers all kinds of unique functions. For example, Lat/Vert mode refers to a scheme in which the channel sum (mono material) and difference (stereo material) were compressed separately in vinyl-disk mastering—still a very useful process even when not cutting vinyl. Similarly, the LA-2A compressor looks simple, but there's more going on under the hood than meets the eye.

## Old Favorites Anew

Recently UA has been shipping models of classic Roland analog processors. My favorite is RE-201 Space Echo ($249), which emulates one of the great tape echoes. UA has modeled the RE-201 right down to the distortion and wow. The plug-in even has a switch for choosing the virtual tape's age, a factor that strongly affected the sound in the original.

The Boss CE-1 chorus pedal ($99) is one of the simplest of all UA plug-ins to operate. The original device was all about thick, shimmering chorus without lots of user parameters. The situation is similar with the Dimension D chorus ($149), which has only four buttons to select a sound. All three Roland models are available individually or in the Roland Classic Series FX Bundle ($399).

## Testing Boundaries

I decided to see how many plug-ins I could run before hitting the UAD-1e's limit, keeping in mind that every plug-in has its own DSP needs. I was able to run seven stereo channels, each with LA-2A and CS-1, and one instance of RealVerb. Another test gave me the same number of channels, each running 1176SE and EQP-1A, and one RealVerb.

On numerous occasions, I requested a plug-in and got a dialog box warning of insufficient DSP and informing me that some plug-ins would be disabled. Unfortunately, it never said which plug-ins were disabled. Because I found no visual indication, I never knew what was really going on. It turns out that communications glitches can occur between the UAD-1e and some hosts and produce "false positives," triggering the dialog box. That kind of uncertainty makes recording engineers extremely nervous in session. UA really should have a clear indication for disabled plug-ins.

You're probably aware that moving audio between a host and a hardware card incurs latency. Many hosts provide latency compensation, but some do not, including Pro Tools LE. UA provides a free DelayComp plug-in for those situations, but it's a bit of work to set everything up.

PRODUCT SUMMARY

UNIVERSAL AUDIO — UAD-1e Expert Pak

DSP card and plug-ins
$1,299

| | |
|---|---|
| FEATURES | 5 |
| EASE OF USE | 4 |
| AUDIO QUALITY | 5 |
| VALUE | 5 |

RATING PRODUCTS FROM 1 TO 5

PROS: Wide variety of plug-ins available, including software models of classic analog processors. Outstanding audio quality. Relieves CPU of DSP burden. Ability to use multiple cards in a system.

CONS: Some rough user-interface edges.

MANUFACTURER
Universal Audio
www.uaudio.com

## Disturbing Graphics

With the exception of RealVerb and DreamVerb, all the UA plug-ins have virtual front panels that resemble the hardware they emulate. In my experience, they look much better than they work—I had to click-and-drag to change stepped switch values on some plug-ins, for heaven's sake. I suppose virtual front panels must really sell products, apparently indicating many users are more concerned with appearance than functionality. I allow some slack when modeling a well-known piece of analog gear, which UA does quite a bit. On the whole, though, I find virtual front panels a flawed idea, and UA's plug-ins fail to sway me on that point.

It is interesting to contrast the reverbs' graphical user interfaces with those of plug-ins that model specific hardware. For example, whereas RealVerb has editable graphics showing the reverb's frequency contours (see **Fig. 4**), neither CS-1's EQ nor the Neve EQ models display equalization curves, let alone editable ones. The lack of graphic displays has always been a difficult drawback of analog equalizers; why perpetuate that paradigm when it is no longer necessary?

## Something for Everyone

To my ears, very few companies do modeling well. The challenge is to emulate specific hardware's sonic behavior and to capture whatever makes the hardware sound pleasing. UA is one of two companies whose models I really love (the other is Line 6). I have yet to hear a model that really sounds as good or better than the best example of the analog hardware it's mimicking, but real vintage units often are not at their best and therefore vary widely in how they sound. UA plug-ins fall just short of the magic I've heard the best analog gear provide, but they're capable of excellent sound that's entirely consistent, easily available, and much, much cheaper than the real thing.

From stompboxes to high-end studio wonders, UA offers a broad selection of plug-ins that make the UAD-1e an extremely versatile tool. Any UAD-1e bundle is a bargain for the fabulous-sounding plug-ins alone, and removing some of the burden from your CPU is a bonus. Aside from feeling like the user interfaces could be made more effective, I love the Expert Pak system. It has become my first call for processing on prominent tracks that really have to sound great.

*Larry the O just returned from Thailand, where he learned to scuba dive and discovered the real meaning of* tropical paradise. *Sawadee krop!*


FIG. 1: The NRV10 is an 8-channel analog mixer that features built-in digital effects and a 10 × 10 FireWire interface.

## M-AUDIO NRV10

# An analog mixer with a flexible FireWire interface.

By Mike Levine

Although many recording musicians have given up traditional consoles in favor of mixing in the box, there is a particular subset of analog mixers that's actually gaining in popularity. I'm referring to those with USB or FireWire interfaces. The number of such products has increased dramatically over the last year or so.

One of the newest entries from that group is the M-Audio NRV10 (see Fig. 1), an 8 × 2 analog mixer equipped with a 10 × 10 FireWire interface that supports audio up to 24-bit, 96 kHz. The unit gives you plenty of signal-routing flexibility, five very serviceable mic pres, built-in effects and EQ, and much more. Although the NRV10 has many live applications (for P.A. and DJ mixing), I'll focus here on its capabilities as a recording mixer.

## Mix and Match

The NRV10 has a small footprint and a standard compact-mixer shape. (Currently there is no rackmount option available.) It features four mono channels (1 to 4), two stereo channels (5/6 and 7/8), and an additional virtual stereo channel (9/10) that exists only as a FireWire output through which the main mix is routed. The physical channel strips are each equipped with short-throw, high-tension faders. Although they feel fairly stiff compared with conventional faders, M-Audio chose them so that they'd be less susceptible to jostling in a club situation. They take a little getting used to, but after that they're fine.

Each channel (except 9/10) also has a basic 3-band EQ: the Low control is fixed at 80 Hz, the Mid at 2.5 kHz, and the High at 12 kHz. The channels each have two aux sends. In the mixer's default routing, aux 1 (Mon) is used primarily for setting up a cue mix. Aux 2 (DFX) sends your signal to the onboard digital effects processor. Two mono aux send jacks with stereo returns let you connect outboard processors. The channel strips also have mute switches, pan controls, trim controls, and a Channel Source button (one of the most important controls on this mixer) that toggles the channel between the analog input and the corresponding FireWire output from the computer.

Channels 1 to 5 have XLR mic inputs and balanced/

unbalanced ¼-inch inputs. Channels 1 to 4 also offer ¼-inch TRS insert points. Stereo channels 5/6 and 7/8 both have two ¼-inch TS mono inputs. Each channel has a mute switch but no solo switch.

The master section features faders to control Phones, Ctrl Room, and Main Mix levels. You also get four return knobs: Aux Rtn 1, Aux Rtn 2, FW 9/10 to Phones, and FW 9/10 to Ctrl Room. The latter two control how much of the main mix coming from the computer goes to the headphones and control room outs, respectively. Also in the master section is a 3-way Phones Source switch that governs what will be heard in the headphone mix. You can choose Main Mix; Monitor, a mix setup using the Aux 1 knobs (which can be used for setting up a separate monitor mix for live talent); and Cue, which lets you monitor any muted channel through the headphones without disturbing the main mix.

For metering, you get a 13-step stereo LED meter with clipping indicators, which shows you the level of the main mix. The individual channels each have peak and mute-status indicators.

## Back It Up

As you would expect, the rear panel offers an assortment of I/O (see **Fig. 2**). You get a pair of FireWire jacks, which is handy because it allows you to daisy-chain other FireWire devices. (M-Audio thoughtfully includes two different types of FireWire cables along with the mixer: 6-pin to 6-pin, and 6-pin to 4-pin.) You also get both ¼-inch and XLR main outputs, ¼-inch control room outputs, ¼-inch inserts for the main outputs (so you can patch in, say, a compressor), a global 48V phantom power switch, and a Kensington Lock Port for locking the unit up with a Kensington security cable.


**FIG. 2: The rear-panel I/O includes ¼-inch and XLR main outputs.**

Also on the rear panel are the power switch and an input for the 12 VDC adapter. The NRV10 is not FireWire bus powered. Like other products of its type, it has no MIDI I/O. To send MIDI into or out of your

REVIE

sequencer, you'll need a separate MIDI interface.

## Soft Sense

The installer disc for the NRV10 loads two pieces of software onto your computer. The Control Panel (see **Fig. 3**) is a basic but useful utility that shows you the levels of all the inputs and outputs on the NRV10. It came in very handy when figuring out routing options.

The other application, NRV10 interFX, was developed jointly by M-Audio and Audiffex. It was designed primarily for hosting effects plug-ins for live use. You can get similar effects-hosting functionality through your digital audio sequencer, so NRV10 interFX isn't necessary in the studio.

Briefly, NRV10 interFX gives you gain control, a compressor, and a gate for each mixer channel, and it lets you insert VST plug-ins into the path of any analog input channel coming through the FireWire bus from the NRV10. So if you were using the NRV10 live, you could access any VST plugs that were on your computer for use on your live sources. NRV10 interFX also comes with six basic but solid VST plug-in effects from Audiffex: a delay, a chorus, a compressor, a flanger, distortion, and an expander/gate.

The NRV10 is compatible with a range of digital audio software, including Digidesign Pro Tools M-Powered. M-Audio includes a Pro Tools M-Powered demo (Mac/Win) with the unit.

## It's the Process

The NRV10 also has its own built-in digital effects, which can be used anytime, including when the mixer is in standalone mode. You can even use them—through the DFX send—on a channel that's returning to the mixer through the FireWire bus.

You get 16 different effects, including several types of reverb, a chorus, a flanger, a mono and a stereo delay, and a couple of combination effects. You can have only one effect active at a time.

The effects sound pretty good, but they're not editable; you get one knob to choose the effect type, and another to select a variation of that effect type (for example, for the hall reverb setting, you can choose from a variety of decay times). Essentially, they're just a collection of presets, which limits their utility in the studio. Probably the biggest benefit of these effects is the ability to add them to the monitor mix when using the mixer's direct monitoring facilities.

## Computer Whiz

Although the NRV10 does function as a standalone analog mixer, if you don't use it with its FireWire I/O and a computer, you won't get the most from it, especially in the studio. Used in conjunction with a digital audio sequencer, it provides a lot of options. For a tracking session, you can bring up to six sources (four mono and two stereo) into your DAW, with their Channel Source buttons set on Channel, and monitor them directly with the built-in effects with zero latency.

Alternatively, you could monitor

PRODUCT SUMMARY

**M-AUDIO NRV10**

analog mixer with FireWire interface
$899.95

| | |
|---|---|
| FEATURES | 4 |
| EASE OF USE | 3 |
| AUDIO QUALITY | 4 |
| VALUE | 3 |

RATING PRODUCTS FROM 1 TO 5

**PROS:** Flexible signal routing. Good-sounding mic pres. Three different monitoring modes. Solid sound quality. XLR and ¼-inch main outputs.

**CONS:** EQ very basic. No editing of built-in effects.

MANUFACTURER
M-Audio
www.m-audio.com

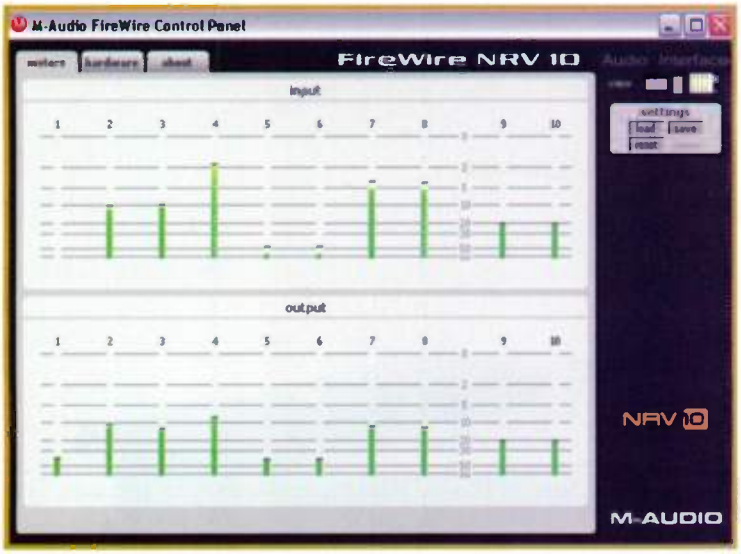



FIG. 3: You can easily see what's going on in the FireWire interface with the included Control Panel software.

through your computer, and use your sequencer's plug-in effects by setting the Channel Source buttons to FireWire. If your record buffer is low enough, latency won't be an issue. You could even monitor some inputs through FireWire and other inputs directly if you wanted. This routing flexibility sets the NRV10 apart from similar products. It lets you route individual channels back through the mixer for monitoring or additional processing, whereas other analog mixers with FireWire interfaces let you send back only a stereo pair.

If you're doing your final mix in the box, simply set the output of your tracks to the NRV10's FireWire 9/10 bus. You'll be able to monitor your mix through the FW 9/10 to Phones and FW 9/10 to Ctrl Room returns. When you're satisfied with how it sounds, bounce it to disk as usual.

If you're running low on CPU power during your mix, you can utilize the NRV10's built-in processing on some or all of your tracks. But this will necessitate sending all of your tracks through the NRV10's physical input channels, which means that your signal must pass through an additional digital-to-analog and analog-to-digital conversion as it's mixed.

## Testing, One, Two

I tested out the NRV10 in tracking, overdubbing, and mixing situations. Once I got the hang of its architecture, I found it to be a very capable and flexible device.

I tested the mic pres on both acoustic guitar (see **Web Clip 1**) and vocals, and found them to be clean sounding with a nice sheen in the high end. According to M-Audio, they're the same pres that are in its Octane preamp (see the review in the December 2004 issue of EM, available online at www.emusician.com).

## The NRV of You

The NRV10 is an appealing product. It combines the features of a mixer and an audio interface, and it sounds really good. Having the choice to monitor the inputs either directly or after the computer's processing gives you a lot of flexibility. The built-in effects and EQ are basic but can be very useful during tracking and overdubbing.

If you're looking for a versatile audio interface but don't want to give up the hands-on aspects of a physical mixing console, you'll definitely want to check out the NRV10.

EM WEB CLIPS

*Mike Levine is an EM senior editor.*


FIG. 1: The Akai EWI4000s includes a built-in analog-style synth, MIDI In and Out ports, and more user-definable messages than ever.

## AKAI PROFESSIONAL EWI4000s

# The latest generation of the EWI is the best yet.

By Scott Wilkinson

As far back as I can remember, I've had a natural ability to pick up just about any wind instrument and get a reasonable sound out of it, even if it's an instrument I've never seen before. But my brain balks at anything with keys or strings. Thus, I've always considered wind-instrument synth controllers to be my ticket to electronic-music performance. I've owned a Lyricon, all three models of Yamaha's WX controllers, and the Akai EWI 1000, the first commercial generation of Nile Steiner's Electronic Wind Instrument.

Akai has continued to refine the EWI since the 1000, introducing several models with new features and refinements. Now comes the EWI4000s, which promises to be the best EWI yet (see Fig. 1).

## Warming Up

The EWI4000s implements a number of useful refinements over previous generations. For example, there is now a Glide Strip running alongside the octave rollers. By default, touching the strip sends Portamento On, but it can be assigned to send any Control Change message from 0 to 99.

Speaking of MIDI messages, the EWI4000s provides more user-definable messages than ever before. You can enable several different messages to be sent simultaneously in response to breath pressure and bite pressure on the mouthpiece. Continuous controllers 0 to 99 can be assigned to the Octave and Hold buttons (sent with a preset value) as well as the Pitch Bend Up and Down plates, which can send varying values depending on how much of your right thumb is in contact with the plates. You can even send Program Change messages by fingering different notes, a feature that Michael Brecker put to good use.

The EWI4000s offers several different fingering modes, including the regular EWI mode and a saxophone mode that more closely mimics a normal sax. In fact, this mode even lets you rest your left pinkie on the G-sharp key just like many sax players do; in this case, the only notes affected by the G-sharp key are G and C.

Also provided are two EVI (Electronic Valve Instrument) fingering modes, which let you finger the main right-hand keys like a valve instrument (such as a trumpet or tuba), shifting registers with the left index finger. I play many brass instruments as well as woodwinds, but I've never been able to wrap my brain around EVI fingering, so I opted to use the EWI and sax modes.


FIG. 2: UniQuest's well-organized editor window provides access to all synth controls.

## Pièce de Résistance

Without question, the most important new feature in the EWI4000s is its internal synthesizer. With all previous models, the synth was an external unit that connected to the instrument with a special cable. This was like an anchor, tying the player to a relatively small area on the stage.

With a self-contained internal synth and AA battery power, the EWI4000s is finally free to roam. Of course, you must still connect its ¼-inch audio output to a sound system, but with a wireless transmitter on your belt and a receiver near the mixer or amp, you can move anywhere within the system's RF range. (If you want to control another synth via MIDI, you'll have to connect a cable from the EWI's MIDI Out to the synth's MIDI In, but this can be done wirelessly as well.)

As in previous generations, the synth is a 2-oscillator subtractive design—quaint by today's megasynth standards, but refreshing in its simplicity. Each oscillator can produce sawtooth, triangle, and/or pulse waveforms with independent level controls, and the pulse width can be modulated at a user-specified frequency and depth. Breath-controlled functions include level and pitch modulation as well as oscillator onset, and a crossfade function follows breath pressure to crossfade between the two oscillators.

The oscillators are mixed before being sent to the dual filter. The multimode filter can be modulated by breath, LFO, and/or a sweep function, and the filters can be linked to double the cutoff slope. A separate formant filter can apply woodwind or string formants, and a noise generator has its own pair of multimode filters with the same parameters as the oscillator filters.

The synth also includes three effects: delay, reverb, and chorus, the last of which provides separate controls for each oscillator. Rounding out the synth are several global parameters, including key trigger, bend range, and vibrato amount for both pitch and amplitude.

With two voices, the synth can play two notes at a time. This allows the EWI to hold one note while playing others or to play two parallel lines separated by any interval.

## Ed-Lib Vibe

The older outboard synth module did have one advantage: all of its parameter controls were on the front panel and immediately available. Now the only way to program the synth is with the included editor-librarian from Sound Quest. Dubbed UniQuest for Akai EWI4000s, the software comes on a CD-ROM with versions for Windows XP and Mac OS X.

To use the editor-librarian, you connect MIDI In and Out from the EWI to your computer's MIDI interface and launch the program. As with most such programs, this one has a library window that displays all 100 presets in a given bank. Clicking on a sound in the bank activates that sound for playing on the EWI, which is very convenient when browsing for sounds. In addition, individual sounds can be dragged-and-dropped between banks.

The editor window is a well-organized, onscreen control panel for the synth (see Fig. 2), and you can hear the results of any tweaks immediately. However, the controls are slow to respond to dragging with the mouse, making it difficult to fine-tune your adjustments.

Also available is a basic 16-track MIDI sequencer (see Fig. 3) and a MIDI monitor, which displays all MIDI messages passing between the EWI and the program.

Unfortunately, I had some problems using the software. First, getting the factory bank from the EWI into the program failed due to SysEx errors. Also, the program wouldn't recognize the bank of sounds I got from Patchman Music (see the sidebar "Patchman to the Rescue"). I was finally able to get those sounds into the EWI using a function that sends SysEx data directly from a disk to the instrument, after which getting a bank from the instrument into the program started to work. Obviously, there are some bugs in the software that Sound Quest should iron out.

## Shut Up and Play

The EWI4000s feels quite hefty, bespeaking quality construction. It cannot be easily played without a

PRODUCT SUMMARY

AKAI PROFESSIONAL EWI4000s

MIDI wind controller
$999

| | |
|---|---|
| FEATURES | 4 |
| EASE OF USE | 3 |
| QUALITY OF SOUNDS | 3 |
| VALUE | 4 |

RATING PRODUCTS FROM 1 TO 5

PROS: Self-contained synth with lightning-fast response. Solid feel. Much less expensive than WX5/VL-70m package.

CONS: Confusing user interface. Factory sounds are mostly uninspiring. No WX fingering mode.

MANUFACTURER
Akai Professional
www.akaipro.com

neck strap because both thumbs must be free to move (left thumb on the octave rollers, right thumb on the pitch-bend plates). Unlike the EWI 1000, this model allows air to pass through it, making it feel more like a real wind instrument.

The user interface consists of two 7-segment LED characters on the underside of the instrument. Though they can be quite cryptic, they're better than no display at all, as on the Yamaha WX instruments. Many of the controls are multifunction buttons, which adds to the initial confusion, but I got used to them eventually.

Playing the built-in synth is a joy thanks to its incredibly fast response. According to Akai, the connection between the controller and synth is 14 times faster than MIDI, making it feel like you're playing an acoustic instrument. The synth is so responsive that singing while playing evokes a ring modulator effect, and flutter-tonguing works like a charm. The Key Delay parameter helps reduce glitch notes caused by imprecise fingering; the default value is 7, but I found that 10 worked best for me without slowing things down appreciably.

As much as I love the response of the internal synth, I am not so enamored of the factory presets, which start to sound alike after a while. Of course, there are some good ones. My favorites include Judd4000 (a classic Lyle Mays lead sound), Stonehenge (a haunting, flutelike sound with wind noise), WoodNGlue (a chorused double pulse-width modulated sound), and NewWood (a clean clarinet type of sound). Patchman Music offers an alternate bank of sounds that I like much better overall.

## PATCHMAN TO THE RESCUE

Patchman Music (www.patchmanmusic.com) specializes in creating synth programs for wind controllers, so it's no wonder the company offers a bank of sounds for the Akai EWI4000s. After some initial problems loading these sounds into the EWI, I finally got the sounds installed and started playing.

What a revelation—each and every sound is beautiful and very responsive. Some are acoustic-instrument simulations, such as Flute, Oboe, and Clarinet, that are more believable than their factory counterparts. (The brass instruments are less realistic than the woodwinds, but brasses are particularly difficult to synthesize.) Of course, they won't fool anyone into thinking they're the real thing, but they are wonderful sounds in their own right.

Then there are the purely synth sounds, such as Blow It Out, D-50 Lead, and Funkatron, which sing their electronic song with exquisite response to breath control. Also included are several excellent synth-bass sounds. Many of the Patchman sounds use the synth's crossfade and breath-activated oscillator-onset functions to vary the sound in a completely natural and—dare I say it?—organic way.

According to Patchman Music founder Matt Traum, the UniQuest editor's slow response to mouse manipulation made it difficult to tweak the sounds to his satisfaction, requiring more time and patience than usual. But it was well worth the effort. If you have an EWI4000s, you owe it to yourself to get a copy of these sounds. For $90, you won't be sorry.

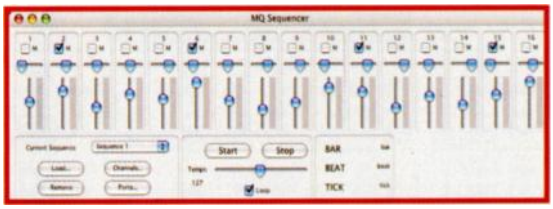



FIG. 3: UniQuest also offers a simple 16-track MIDI sequencer.

Controlling a MIDI sound module feels slightly slower than using the internal synth. I used a Yamaha VL-1m sound module, which is designed for wind control, and I disabled Aftertouch from being sent in response to breath pressure to reduce the amount of transmitted data. Still, it felt just a bit sluggish; for example, some notes in fast scales seemed to get lost via MIDI, whereas they were clearly heard from the internal synth. Decreasing the Key Delay definitely helped, but it also increased the incidence of glitch notes. To be fair, this is a relatively minor difference that should not pose a significant problem.

Compared with the WX5, the EWI4000s is much heavier, and its user interface is better. Aside from the 2-character display, the onboard breath-sensitivity and other physical controls are small knobs that are much easier to manipulate than the set screws and DIP switches on the WX5.

As for the difference between playing the WX5, with its moving keys, and the EWI, with its nonmoving touch-sensitive keys, that's a matter of personal preference. The WX5 feels more like a sax in this regard, while the EWI feels more like a recorder. The same goes for the mouthpiece—the WX5 mouthpiece looks and feels just like a sax mouthpiece, complete with a "reed" that invokes Pitch Bend and other MIDI messages, while the EWI has a hard plastic tube. I definitely prefer the EWI's octave rollers to the WX5's nested octave keys.

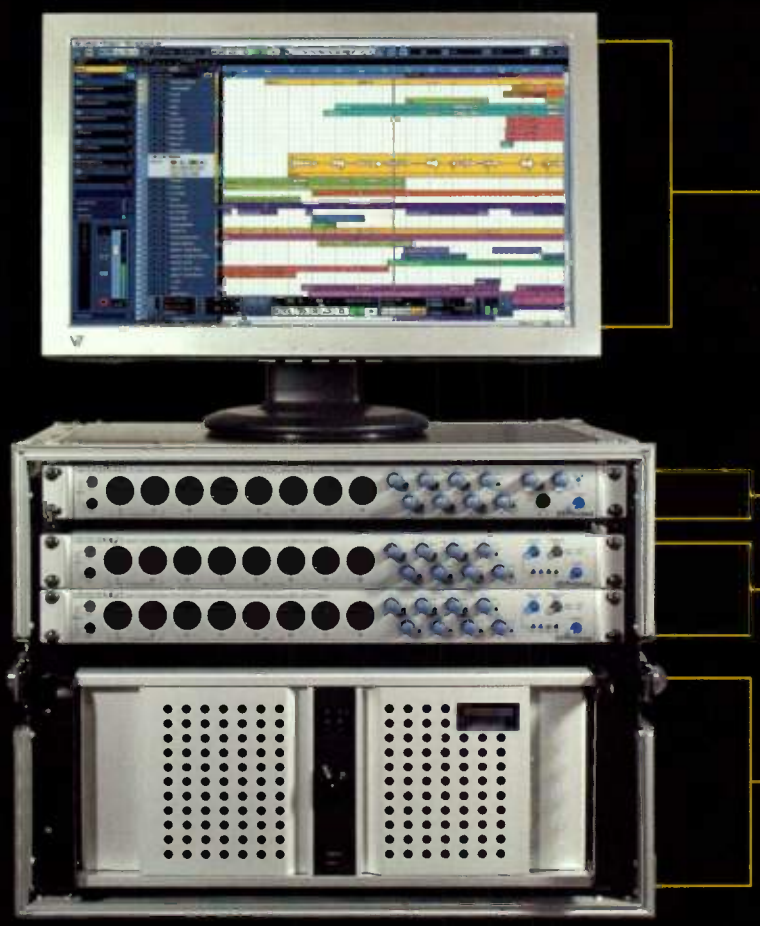
On the downside, I found myself missing the WX5's fingering pattern, which lets you play an octave and a half without shifting octave keys. I kept doing this on the EWI and getting an unexpected note. I wish the EWI had a WX fingering mode, though that's probably not possible due to corporate competition and patents.

## Bottom Line

Integrating the sound module into the body of the EWI4000s is a big step forward for wind controllers. With the appropriate wireless system, wind players can now freely move around the stage. When you consider that the WX5 with Yamaha's VL-70m sound module lists for more than twice the price of the Akai, the EWI4000s is a very attractive package indeed.

*Former EM technical editor Scott Wilkinson loves to make music by blowing.*

# ROAD-READY



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PreSonus

## FIRESTUDIO

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- DSP matrix routing mixer
- Word clock I/O, MIDI I/O, SPDIF I/O

PreSonus

## DIGIMAX FS

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(2 units shown for an additional 16 channels)

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- JetPLL jitter elimination technology
- Direct outputs, insert points, DAC inputs

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# EM Marketplace


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When Jars of Clay hit the road to promote their current release GOOD MONSTERS, they wanted to record their shows in hopes of releasing a live recording after the tour. After recording the first few shows, they realized, not only was their Creation Station system rock solid, but that the recordings were so high quality they decided to release a live record, called LIVE MONSTERS, during the tour! How did they do it?

It starts with a recording system capable of dealing with the demands of a national tour. At the heart of the system is the Sweetwater Creation Station – designed and configured by the computer audio experts at Sweetwater specifically for audio production. Connect the PreSonus FireStudio™ and two DigiMax FS® for 24 Class A microphone preamplifiers with Jet PLL synchronization technology enabling hours of rock-solid continuous multi-channel recording. Finally, add Steinberg's Cubase 4 - the most advanced audio recording and production software with the ability to mix during recording, send sequencer and virtual instrument tracks in real time and more, for a fully integrated hardware and software recording rig that can handle anything thrown at it.

"Our recording rig is exactly what we need it to be: dependable, easy-to-use, and powerful," says Jars of Clay guitarist Matt Odmark. "It has become an integral part of our production and recording. The PreSonus preamps sound amazing, Cubase 4 is unbelievably creative and powerful and the Sweetwater Creation Station never skips a beat, literally. Thanks Sweetwater for the sweetest recording rig!"

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Music Instruments & Pro Audio

## FXPANSION

### BFD Percussion

By Marty Cutler

EM ★★★★★ hot pick

FXpansion's BFD is a software drum module that relies on extensively layered samples to provide hyperrealistic, extremely expressive drum kits and performances. Until recently, BFD expansion packs contained sampled pieces of drum kits (called Kit-Pieces) almost exclusively, but fortunately, the company has turned its attention toward percussion instruments. With the release of *BFD Percussion* ($249), the results are impressive.

To host the samples in *BFD Percussion*, you'll need BFD 1.5.45 (a free update) or later. An advantage of version 1.5 is the addition of a slot in which you can load a traditional kit, additional percussion, or any combination, up to a total of 18 Kit-Pieces. Although BFD is not a drum machine in the strictest sense, it hosts MIDI files that you can sequence and trigger with notes as you would with sampled loops. (For more details on BFD's inner workings, see the May 2004 issue of EM, available online at www.emusician.com.)

*BFD Percussion*'s two DVDs furnish 26 GB of beautifully sampled 24-bit, 44.1 kHz instruments ranging from conventional hand percussion to found objects and appliances. All the samples were recorded at Maryland's Omega Studios under the production guidance of John Emrich, who played all the instruments. Instruments comprise multiple key assignments, each with Velocity-switched sample maps as many as 46 layers deep, offering excellent control over dynamics and performances that sound startlingly organic. Samples recorded with room and PZM microphones add to the realism, and you can vary the degree of ambience for each Kit-Piece.

You can elect to install either the entire 26 GB sound set or medium or small installations, which supply as many as 24 and 16 Velocity layers, respectively. I auditioned all three sizes on my dual-processor 1.42 GHz Power Mac G4 with Mac OS X 10.4.8. Installation was simple, requiring only my administrator's password and the serial number provided with the package. You can easily install a smaller or larger set as your system resources warrant.

#### Percs of the Job

*BFD Percussion* gives you an exceedingly rich menu of traditional instruments, including Latin percussion such as bongos, congas, timbales, and guiro; Turkish instruments such as tar; African djembe, shekere, and udu; Balinese nipple gong; Arabian darbuka and riq; and Pakistani tambourine. Although I found no instruments from India, I hope that means FXpansion has more expansion packs in the wings.

The sounds are consistently well recorded, and the programming—especially the Velocity layering—is smooth and articulate. Once installed, preset kits often mingle percussion from different parts of the globe with nontraditional instruments. For example, ChaCha_118 adds a Rubbermaid trash can with bongos and other Latin percussion, and Mambo2_200 offers a percussive stew of djembe, bongos, temple block, guiro, cowbell, and other typical instruments with a saw blade and coffee can. Other sounds include shoe taps, wooden and cardboard boxes, frying pans, and the kitchen sink (literally), sampled with multiple strike zones and playing techniques, including striking with mallets.

Accompanying MIDI files stored in BFD's built-in Groove Librarian pair with the kits, providing a wealth of terrific grooves. For example, GTron grooves like crazy, merging a busy, second-line drumming feel with djembe, tar, goatherder bells on a stick, and a box of unknown origin (see **Web Clip 1**). The most startling aspect of the groove-and-sample pairing is their organic nature; to my ear, they are often indistinguishable from high-quality sampled grooves. Other MIDI files have no associated percussion kits, but matching kits and grooves at random proved quite effective (see **Web Clip 2**). You can also create custom kits and import your own MIDI grooves into the librarian.

The excellent booklet that comes with *BFD Percussion* furnishes information on each Kit-Piece, revealing its global origin, material, model, and the articulations provided. It also lists file sizes, making it easy to calculate the amount of RAM required (BFD can also stream from disk).

*BFD Percussion* is not inexpensive, and you'll need to buy the BFD application if you don't already own it. In its richness of sound and expressive capability, however, *BFD Percussion* is head and shoulders above any percussion or drum module (hardware or software) I have heard. I recommend it highly to anyone searching for a great-sounding set of percussive tools to fuel their creativity.

EM WEB CLIPS

**Value (1 through 5): 5**
FXpansion
www.fxpansion.com


**FXpansion *BFD Percussion* delivers an expressive set of percussive instruments selected from all parts of the globe, including your toolshed and kitchen.**

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## ECHO DIGITAL AUDIO

### AudioFire4 (Mac/Win)

By Vijith Assar

The AudioFire4 ($299), little brother of the successful AudioFire8 and AudioFire12, is Echo Digital Audio's latest offering in FireWire audio interfaces. It supports a wide range of input formats, and you get both balanced and unbalanced inputs and outputs. The AudioFire4 handles 16- and 24-bit audio at 44.1, 48, 88.2, or 96 kHz sampling rates.

I wasn't able to try the AudioFire4 with my ancient Windows 2000 desktop PC, but the unit supports Windows XP and Vista 32-bit, with 64-bit support on the way. I did test it on a wide variety of G3, G4, and Intel Macs, and it worked quite well on most of them. However, on a dual G4 Power Mac, playback gave me only an odd clicking noise. I emailed Echo, and within a couple of hours the company sent me a newer version of the I/O routing console, which automatically flashed the device with updated firmware that solved the problem.

#### The Ins and Outs

The AudioFire4 rear panel holds S/PDIF and MIDI jacks (for the obligatory 1-port MIDI interface), two line inputs, and four line outputs. The two 6-pin FireWire ports support device chaining and bus-powered operation, but you also get a power supply for use with laptops having 4-pin FireWire connectors. The front panel holds two combo-jack inputs and a headphone output, each with a dedicated level control. Input switching is handled by a hardware sensor; for the AudioFire4, an XLR plug always means mic level.


The AudioFire4 front panel sports universal input and headphone output jacks and associated level controls. The rear panel holds all other I/O.

Touching the input trim knobs sometimes caused loud pops through the outputs, suggesting questionable isolation. Phantom power added a little noise, but other than that, the preamps were surprisingly clean. I didn't hear any problematic buzzes or hisses until I cranked the gain to the highest levels.

Mackie Tracktion is included as a free recording platform, but I was surprised to see that it was version 1. Echo says that Tracktion 2 will be included as soon as the Vista version is available.

The routing-control program for the AudioFire4 was developed in-house, and it is excellent. You can link the input signals for stereo operation as necessary and then mirror them to any of the outputs, including the S/PDIF jacks, for latency-free monitoring. In addition, levels for playback and live monitoring can be adjusted separately, and you can even set them differently for each output pair. The only shortcomings are that phantom power isn't individually switchable and the black-on-black color scheme sometimes makes buttons hard to discern, but at least the text or numerical value is always visible, even when the button around it is not.

#### The Soft Touch

The AudioFire4 is a very capable unit, but many of its functions can be accessed only through software. Fortunately, Echo has its software game together, but some additional hardware controls would be welcome. The missing hardware switch for phantom power might be a problem. Lack of an output volume control could be an issue for anyone who plans to plug directly into a pair of powered monitors. The unit would be considerably compromised if support or software updates were to lapse, but Echo has a strong track record of legacy support, and relying on software controls does help keep the cost down. If you're planning to pair the AudioFire4 with a small mixer, the missing hardware controls will be supplanted by the mixer's, and I can recommend the AudioFire4 without hesitation.

I was surprised to find a standalone mode in such an inexpensive device. Every time you quit the I/O configuration console, your most recent settings are saved to the AudioFire4's onboard memory. That lets you use it without a computer for basic mixing, routing, and monitoring tasks. Even in a worst-case obsolescence scenario, you could program the unit from another machine and then move it into your recording rig until you need to change the setup again.

The AudioFire4 gives Echo access to the lower end of the market, but there's a lot of competition in inexpensive audio interfaces. Although not remarkable, the AudioFire4 is a solid player and is fairly priced. Having a selection of products as cheap and powerful as this to choose from is pretty incredible.

**Value (1 through 5): 3**
Echo Digital Audio
www.echoaudio.com

## WAVEMACHINE LABS

### Drumagog Pro 4 (Mac/Win)

By rachMiel

Drumagog is a drum replacer, an effects plug-in that automatically replaces audio drumbeats with samples of your choice. Say the kick track in your latest song is too loose and flabby. Just feed the audio to Drumagog and replace the flab with new, tight kicks from Drumagog's built-in sample library or your own personal storehouse.

Drumagog comes in four flavors: Basic (limited functionality, $199), BFD (limited functionality and BFD compatible, $199), Pro (full functionality, $289), and Platinum (full functionality and BFD compatible, $379). I'd strongly recommend going with the Pro or Platinum version because many nontrivial features are missing from the limited versions: variable pitch, MIDI support, autoducking, advanced triggering modes, and so on.

On the PC side, Drumagog requires Windows 98 or later and will run as a VST or Digidesign Pro Tools LE or TDM plug-in. On the Mac side, it will


**Drumagog's Main page shows replacement samples on the left and basic controls on the right. In many cases, that's all you need.**

run under Mac OS X as a VST, AU, or Pro Tools LE or TDM plug-in. Under OS 9 it will work only as a VST plug-in. Each instance of Drumagog took a bit more than 4 percent of the CPU on my 2.33 GHz Pentium 4 PC.

### In Practice

Drumagog is easy enough to use. First you must segregate all your drum parts so that you have only one part per track. You then insert an instance of Drumagog as an effects plug-in on each track and choose a replacement sample, and you're in business. Often Drumagog's default settings require no further adjustment. But if they do—for example, if some of the original beats fail to trigger crisp replacements—you can fine-tune the sensitivity and resolution using an intuitive graphical *x-y* pad.

Drumagog's controls are presented in three pages: Main, Samples, and Advanced. Main is home base; you use it to choose replacement samples as well as to set levels, triggering sensitivity, resolution, filtering, drum hit position, blend, pitch, and onboard synth parameters. If your project is clear-cut, Main may well be the only Drumagog page you'll have to visit. The Samples page is where you manage (view, create, edit, and audition) sample maps. You use the Advanced page to set specialized parameters such as latency, MIDI I/O, autoducking, dynamic tracking, stealth response and crossfade, and automatic sampling-rate conversion.

Most of the drum replacement controls are standard, but a few deserve special comment. Drumagog supports three kinds of multisampling, and the capabilities are fabulous. You can trigger dynamic multisamples (differentiated by volume), random multisamples, and positional multisamples (differentiated by hit position). You can also mimic beats played by the left and right hands. Stealth mode lets audio below a set threshold pass through unchanged; you can use that to preserve parts of a track while replacing other parts. Finally, autoducking lets you get rid of bleed-through, in which one part is audible on another part's track.

### Pros and Cons

Drumagog has a lot going for it. The GUI is well thought out, and details like the graphical *x-y* pad are user friendly. I like that you can easily load your own samples and create custom dynamic, positional, or random multisample maps. And I really like how it scales in complexity according to the user's needs. Beginners can plug and play. Intermediate users can avail themselves of sophisticated features like multisampling. And professionals can use the advanced triggering controls to ensure that each beat is replaced flawlessly.

Of course, there is room for improvement. There is no undo and no pop-up help, both of which are sorely missed. Timing was fine in Cakewalk Sonar, because it supports plug-in delay compensation, but in Sony Acid, I used the nonfixed-latency version of Drumagog, and timing was an issue. For the most part, samples loaded quickly, but sometimes they took surprisingly long (five to ten seconds). And I found the factory library of replacement drum samples a bit dull.

If you do a fair amount of drum replacement therapy, Drumagog Pro is a must-have, or at least a must-try. (You can download a 14-day trial version.) Though not the only game in town, Drumagog is more full featured than most, offering much finer control of triggering and multisampling. And it has cool extras such as an onboard synth to add spice to your beats; stealth mode; and dynamic, random, and positional multisampling. If you want to breathe new vitality into old audio drum tracks, Drumagog is a great way to go (see **Web Clip 1**).

EM WEB CLIPS

**Value (1 through 5): 4**
WaveMachine Labs
www.drumagog.com

## KARMA AUDIO

## K6 Ribbon Microphone

By Eli Crews

The low cost of manufacturing in China has allowed mic designers in the United States to start up with much less capital than in the past. The result has been an ever-expanding supply of inexpensive mics. Karma Audio, which has its own factory in Shanghai, has joined the fray of companies providing low-cost ribbon mics to a new generation of engineers looking for that magical ribbon sound.

### Big for Its Britches

The K6 ($600) has a striking appearance. Its horizontal slots and vertical side fins are reminiscent of the Royer R-121, but the K6 is roughly twice the diameter. Below the grille, the Karma logo is embossed into a black faux-snakeskin sheath, which encases the mic like a glove.

At the base of the K6 is a long threaded stem. At the bottom of that is the XLR input. The purpose of this protuberance is to screw into the included shockmount. As I attempted to do so, two problems arose: first, the thread is so long that it takes quite a bit of effort to screw it on; and second, even when screwed to the hilt, the shockmount still rattles and shakes against the mic. That's the last thing you want from a device intended to decrease vibrations.

### A Unique Voice

The K6 is a phantom-powered ribbon mic, which is certainly atypical. If you're used to working with ribbons, you have to retrain your brain to flip the +48V switch on even though you know a ribbon mic is on the other side of the cable. The advantage of this design is that

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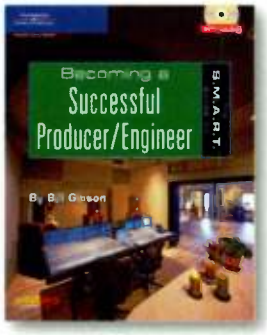


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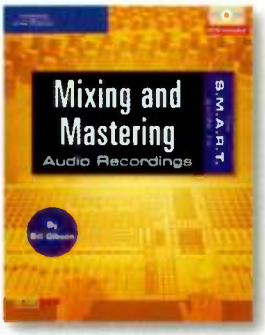


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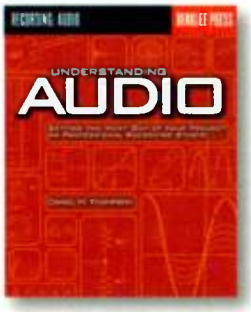


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you don't have to worry about accidentally frying your mic with the phantom power. Although I don't own any other phantom-powered ribbon mics, I compared the K6 to some of the ribbon mics I do possess.

Each of these mics costs several times more than the K6, so it wasn't an apples-to-apples comparison. The Coles 4038, Royer R-121, Royer SF-12, and vintage RCA 77DX each have their own distinct character, and collectively show a wide spectrum of what the "ribbon sound" really is. Each has a different sound, but all are stellar. Whether on horns, acoustic guitar, voice, or drums, the Karma couldn't capture as natural a sound as the others. That didn't surprise me, considering the price difference. (It's important to note that the K6 has a much hotter output than any of these other mics, so unlike the other ribbons listed, it can be used with preamps without tons of gain.)

The K6 had a unique, usable sound on selected sources, but it was too dark and boomy for most applications. One of my favorite uses for it was as a room mic for an electric guitar amp. Its low-end boost and top-end rolloff were perfect for that application. I also had success with it as a secondary mic next to a high-end tube condenser, adding "chest" to a reedy male vocal and some nice "clunk" to a shaker track. I liked it somewhat on flute because it tamed the shrillness of the instrument and enhanced the body, due to its severe drop-off around 5 kHz and bump around 180 Hz (see **Web Clips 1a and 1b**).

**The Karma K6 is a phantom-powered ribbon mic that was designed *not* to sound like the typical ribbon.**

In general, the K6 had kind of a raspy sound. It was a slightly hoarse quality that made my clients say things like "lo-fi" and "gritty," almost as if the acoustic instruments were running through a tube amp with the gain a little too high (see **Web Clips 2a and 2b**). Now, for the right application, that aspect could be really cool, especially if you have other mics to get the pristine sounds. I wouldn't expect to get ultra-high-fidelity recordings through the K6, so I wouldn't recommend it as the mic to start your collection with. Talking to the folks at Karma made it clear they were out to design a unique-sounding mic, not just another ribbon copycat.

### Fringe Benefits

For the money, the K6 is not a bad deal. It comes with a three-year warranty and a sturdy, lightweight flight case with a combination lock. I would recommend that you give the K6 a listen before purchasing, to see whether its character suits your needs. As luck would have it, you can do just that through Karma's unique seven-day trial program. EM WEB CLIPS

**Value (1 through 5): 3**
Karma Audio
www.karmaaudio.com

## LINE 6

## LowDown Studio 110

By Geary Yelton

Line 6 owes its success to realistically modeled guitars and basses, amps, speaker cabinets, and stompboxes. Its POD processors, Variax guitars and basses, and Vetta guitar amps have all contributed to establishing the company as an industry-leading innovator. Last year Line 6 launched the LowDown series of portable combo bass amps, beginning with three models at retail prices from $569 to $839. In November it shipped its most compact and affordable bass amp yet, the LowDown Studio 110 ($349).

EM ★★★★★ hot pick

Designed primarily for studio use, the Studio 110 is a 12.5-inch cube covered in black pile, containing a heavy-duty 10-inch speaker and an amplifier rated at 75W into 4Ω. A metal grille covers the front, and the backplate sports an unbalanced ¼-inch preamp output, a balanced XLR direct output, a ground-lift switch, a power switch, and an IEC power receptacle. On the top panel are a ¼-inch instrument input, four buttons for selecting amp models, seven chrome-plated knobs, a ¼-inch headphone output (which defeats the speaker), and a mini-jack input for an MP3 player, a CD player, or a drum machine. For basses with active electronics, the instrument input has a button that enables a –10 dB pad.

### Bird's-Eye Lowdown

The Studio 110 furnishes five digital models inspired by specific amps and cabinets, with buttons for selecting Clean, R&B, Rock, and Grind. It also simulates a bass synth and provides controls for editing waveform, envelope, and filter parameters. If you change the settings for any preset, its corresponding button flashes; holding the button down for two seconds replaces the preset with a snapshot of the knob positions.

The Clean model emulates an Eden WT550 amplifier with a 4 × 10 cabinet. It has a round, punchy tone suitable for countless situations (see **Web Clip 1**). R&B gets its sound from a 1968 Ampeg B-15 Portaflex, an amp-and-speaker combination popular with Motown session players (see **Web Clip 2**). Rock sounds a lot like a 1974 Ampeg SVT amp with an 8 × 10 cabinet, a classic rock 'n' roll rig that's graced many club and concert stages (see **Web Clip 3**). Grind reproduces an SVT rig overdriven by a Tech 21 SansAmp PSA-1, a rackmount preamp for guitar and bass that's popular for metal, mixed with a direct signal (see **Web Clip 4**).

The fifth model is hidden unless you read the manual. If you hold down the Clean and R&B buttons as you power up the Studio 110, Clean is replaced by the Brit model, which is based on a 1968 Marshall Super Bass (see **Web Clip 5**). Repeating the procedure restores the Clean model.

I plugged in my bass and was immediately pleased with all five amp-and-cabinet models. The top-panel knobs let me dial up changes in tone and sustain. Four knobs control bass, low-mid, high-mid, and treble boost and cut. The Opto Comp knob determines the threshold of the onboard compressor, which simulates the Teletronix LA-2A. The Drive knob functions just like the Gain knob on an Ampeg or Eden amp.

Pressing two buttons simultaneously


**Pro studio bass amps just got smaller.**

turns on the Synth model, which reassigns all knobs except the master volume. One knob controls a lowpass filter's cutoff, and another controls resonance. The Attack/Decay knob affects the filter envelope, and the Envelope knob changes its depth. The Waveform knob yields nine different sounds, some featuring chorus, detuning, and octave effects. The Studio 110's synth timbres range from sustained fuzz to outrageous funk (see **Web Clip 6**).

### Rock Solid

Let's face it, my concert-playing days are behind me. For practice, recording sessions, and casual gigs, I no longer need a hefty bass rig. I want a versatile combo amp that's as compact as possible but still sounds impressive in intimate settings. Other amps might fit the bill, but few are as portable as the Studio 110, and even fewer deliver its tonal range. If I need more output, I can route the Studio 110 to the house system using its XLR direct output, which quite effectively eliminates the need for a direct box.

At 23 pounds, the LowDown Studio 110 is lightweight enough to take nearly anywhere, and it sounds awesome. Granted, it has no footswitch input for changing presets, no tuner, and no onboard effects other than compression; for those, you'll have to spring for one of the pricier LowDowns. Nonetheless, the Studio 110 deserves the highest rating I can give. It is indeed an amazing value, and as good as it gets with current technology. EM

EM WEB CLIPS

**Value (1 through 5): 5**

Line 6

www.line6.com

# EM Marketplace

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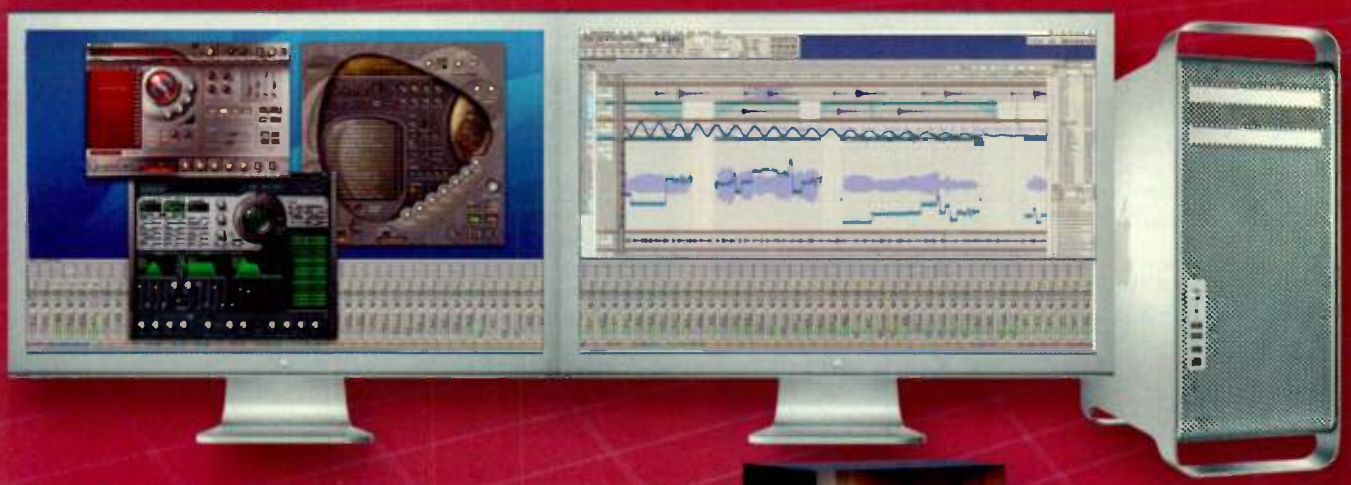

## Intel-Xeon Quad Core Processing

Running on an Apple Mac Pro tower, Digital Performer 5 can play 1,000 real-time plugins—an incredible benchmark that demonstrates just how much native processing power the MOTU desktop studio now delivers. Equip yourself with all the virtual instruments and powerful plug-ins you'll need to sculpt your own, unique sound, such as the unique Ethno Instrument.



## 96-channel PCI Express Audio

The MOTU PCI-424 core system lets you connect up to four interfaces to a single card for up to 48 channels of 192kHz recording and playback (shown) or 96 channels 96kHz recording and playback. Mix and match any combination of three interface models to suite your I/O needs. The two-rack HD192 interface provides 12 XLR in/out with AES/EBU digital I/O (with sample rate conversion), 19-segment front panel metering and an incredible measured signal to noise ratio of 120dB. The 24io offers an astonishing 24 TRS analog inputs and outputs in a single rack space, all at 96kHz. And the 2408mk3 provides 24 channels of ADAT optical (3 banks), 24 channels of TDIF (3 banks), 8 channels of TRS analog, S/PDIF and seamless mixing across all connected interfaces.

## Waves native processing

Waves has long been synonymous with quality plug-ins, and the Waves Platinum Bundle contains a huge range of top-quality Waves processing for your DP5 studio. The Platinum Bundle now includes Waves Tune LT, L3 Ultramaximizer, and IR-L Convolution Reverb as well as all the plug-ins found in the Waves Gold and Masters bundles. Platinum brings extraordinary signal processing power to DP5, for tracking, mixing, mastering, and sound design. From dynamics processing, equalization, and reverb to pitch correction, spatial imaging, and beyond, Waves Platinum Bundle is a must-have for every MOTU studio.


**The MOTU experts at Sweetwater can build the perfect DP5 desktop rig for you. We'll help you select the right components, and we can even install, configure and test the entire system for you. Why shop anywhere else?**

## FilterFreak and SoundToys

Rejoice! The SoundToys Native Effects bundle is now Audio Unit compatible, so DP5 users can experience the incredible sound of EchoBoy, FilterFreak, PhaseMistress Crystalizer, and Tremolator. With the focus on fat analog sounding effects, versatile control and easy of use, SoundToys plugs get you great sounds fast. These are the tools the pros use to make a mix into a hit. Trent Reznor of Nine Inch Nails says, "Calling EchoBoy a delay plug-in is doing it a disservice. It has become the first thing I turn to for treating a wide variety of sources."



Total Workstation Bundle $599
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- SampleTank 2 XL
- Sonik Synth 2
- Miroslav Philharmonik
- Ampeg SVX
- Amplitube 2
- Classik Studio Reverb
- T-RackS

## IK Total Bundle Series

The IK Multimedia Total Bundle Series offers professional quality production tools at prices every musician can afford. IK's Total Studio Bundle with 7 award-winning plug-ins, 90 DSP Effects, 8000 sounds and over 21.5 GB of samples offers a diverse collection of instruments and effects for every mix. The Total Workstation Bundle delivers 3 award-winning virtual instrument workstations covering every style of music and genre, powered by SampleTank's advanced sample technology, built-in DSP, and easy to use interfaces. The Total Effects Bundle includes 4 award-winning effect plug-in suites for guitar, bass, mixing and mastering, all modeled after the most sought-after hardware gear, with 90 ultra accurate, analog modeled DSP emulations. Musicians First.

## On-demand processing

The RECEPTOR PRO from Muse Research is the ideal way to run your favorite plug-ins live, and when in the studio it integrates seamlessly with Digital Performer and adds additional horsepower to your host Mac. Available with 400GB or 750GB drives, you'll love the way RECEPTOR PRO hosts your favorite plug-ins, making it easy to create or find any sound, then playing that sound with world-class sonic quality. Whether you play keyboards, guitar, electronic drums, or use effects plug-ins for mixing, Receptor gives you a stable, convenient, and easy way to run your plug-ins.



## Komplete control

For DP5 users who want it all: Reaktor5, Kontakt2, Guitar Rig 2 software, Absynth4, Battery3, FM8, B4II, Akoustik Piano, Elektrik Piano, Vokator, Spektral Delay and Pro-53 in a unified interface with hands-on control — Native Instruments KOMPLETE 4 and KORE put an infinite universe of sound at your finger tips. Every preset included in NI KOMPLETE 4, more than 8,500 in total, has been preconfigured and categorized in KORE with searchable musical attributes and hands-on controller assignments. This seamless integration of software and hardware turns Native Instrument's award winning synthesizers and samplers into tactile instruments.



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# The MOTU Studio: 1,000 plug-ins and counting

**Keystation Pro 88 features**

- 88-key hammer-action, velocity-sensitive keyboard
- Powered via USB bus (cable included) or optional 9V power supply
- 24 MIDI-assignable rotary controllers
- 22 MIDI-assignable buttons
- 9 MIDI-assignable Alps faders
- MIDI-assignable pitch bend and modulation wheels

## 88 Weighted Hammer-Action Keys

Digital Performer 5 gives you unprecedented control over your MIDI and audio tracks. And what better way to take advantage of this hands-on control than the M-Audio Keystation Pro 88. Regardless of whether you're a seasoned pro or just ready to take your music to the next level, these hammer action keys are so expressive that you just won't want to stop playing! The Pro 88 could easily become your sole keyboard in the studio or onstage. Yet the Keystation Pro 88 weighs only 47 lbs. — half of most weighted-action keyboards! And the Pro 88's extensive features make it the most comprehensive and competitive product of its kind!


## Control room monitoring

The PreSonus Central Station is the missing link between your MOTU recording interface, studio monitors, input sources and the artist. Featuring 5 sets of stereo inputs (3 analog and 2 digital with 192kHz D/A conversion), the Central Station allows you to switch between 3 different sets of studio monitor outputs while maintaining a purely passive signal path. The main audio path uses no amplifier stages including op amps, active IC's or chips. This eliminates coloration, noise and distortion, enabling you to hear your mixes more clearly and minimize ear fatigue. In addition, the Central Station features a complete studio communication solution with built-in condenser talkback microphone, MUTE, DIM, two separate headphone outputs plus a cue output to enhance the creative process. A fast-acting 30 segment LED is also supplied for flawless visual metering of levels both in dBu and dBfs mode. Communicate with the artist via talkback. Send a headphone mix to the artist while listening to the main mix in the control room and more. The Central Station brings all of your inputs and outputs together to work in harmony to enhance the creative music production process.

## Advanced waveform editing

Your DP mastering and processing lab awaits you: BIAS Peak Pro 5 delivers award winning editing and sound design tools, plus the world's very best native mastering solution for Mac OS X. With advanced playlisting. Superb final-stage processing. Disc burning. Plus PQ subcodes, DDP export (optional add on), and other 100% Redbook-compliant features. Need even more power? Check out our Peak Pro XT 5 bundle with over $1,000 worth of additional tools, including our acclaimed SoundSoap Pro, SoundSoap 2 (noise reduction and restoration), Sqweez-3 & 5 (linear phase multiband compression/ limiter/upward expander), Reveal (precision analysis suite), PitchCraft (super natural pitch correction/ transformation), Repli-Q (linear phase EQ matching), SuperFreq (4,6,8, & 10 band parametric EQ) and GateEx (advanced noise gate with downward expander) — all at an amazing price. So, when you're ready to master, Peak Pro 5 has everything you need. It's the perfect complement — and finishing touch — to Digital Performer 5.




## Professional pad controller

The Akai Professional MPD24 is the velocity sensitive pad controller for musicians and DJs working with sampled sounds. The MPD24 features 16 MPC-style velocity and pressure sensitive pads plus transport controls for interfacing with Digital Performer and your virtual instruments. You get Akai's exclusive feel: either MPC 16 Levels or Full Level features for ultimate pad control. Now add four selectable pad banks totaling 64 pads, six assignable faders and eight assignable and 360 degree knobs for transmitting MIDI Control Change data. Included editor/librarian software gives you complete, intuitive programming and control for DP5 all of your other software titles. The MPD24 provides unprecedented creative freedom for manipulating sampled material.

**The MOTU experts at Sweetwater can build the perfect DP5 desktop rig for you. We'll help you select the right components, and we can even install, configure and test the entire system for you. Why shop anywhere else?**



## Accurate monitoring

The Mackie HR-Series Active Studio Monitors are considered some of the most loved and trusted nearfield studio monitors of all time, and with good reason. These award-winning bi-amplified monitors offer a performance that rivals monitors costing two or three times their price. Namely, a stereo field that's wide, deep and incredibly detailed. Low frequencies that are no more or less than what you've recorded. High and mid-range frequencies that are clean and articulated. Plus the sweetest of sweet spots. Whether it's the 6-inch HR-624, 8-inch HR-824 or dual 6-inch 626, there's an HR Series monitor that will tell you the truth, the whole truth, and nothing but the truth.

## New hands-on control for DP5

The new Mackie Control Universal Pro control surface gives you ultimate hands-on control of your Digital Performer desktop studio. Nine motorized, touch-sensitive Penny + Giles faders, eight V-Pots and more than 50 master buttons let you tweak parameters to your heart's content. Unlike generic MIDI controllers, the MCU Pro employs a sophisticated communication protocol that delivers ultra-precise control, makes setup easy - no mapping required - and enables you to see your mix in action with real-time visual feedback via the huge backlit LCD and eight LED rings. Apply the custom overlay for Digital Performer for dedicated labeling of DP-specific functions. The MCU Pro is the ultimate way to mix in DP5!

## Power conditioning

The Monster Power Pro 900 is designed for high-performance hookup of digital and analog components to AC power for all your MOTU studio equipment, providing optimized Monster Clean Power and surge protection for AC power lines. Advanced features include Monster's unique Clean Power Stage 1 filtering for high quality sound and Dual Mode Plus protection with audible alarm for maximum protection and performance. For even more complete protection, the Power Pro 5100 features Monster's Clean Power filtering, color-coded outlets, audible and visual indicators for ground and protection status monitoring, extra-long high current Monster PowerLine cords for optimum power delivery, 24k gold plated contacts on grounded plug for maximum conductivity, 12 programmable outlets, a digital volt meter, Clean Power Stage 4 filtering, 5 filters, sequenced AC power on/off, 3145 joule rating, built-in rack mounts and handles. Get Monster Power today.

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Music Instruments & Pro Audio

# Tracking in Soulville

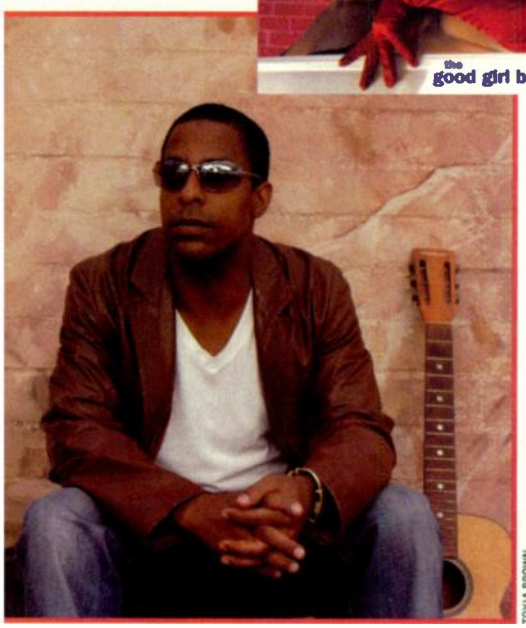
By Diane Gershuny

## The making of the Soul of John Black's latest CD.

This is drinking music," says John "JB" Bigham, front man for the Soul of John Black. "I wanted to make down-home-blues party music," he says of the band's latest CD, *The Good Girl Blues* (Cadabra Records, 2007). You can put it on, cook up some food, and play cards!"

That stylistic approach differs from the band's self-titled debut release in 2003 that blended soul, hip-hop, funk, and Afro-beat influences. That album received raves from the *Wall Street Journal, Interview, Blender, Billboard,* the *New Yorker,* and *Rolling Stone.*

Guitarist, keyboardist, and songwriter Bigham played for eight years in the funk-ska-punk band Fishbone before embarking on his solo career. He also had stage and studio stints with a range of artists including Eminem, Dr Dre, Nikka Costa, and Bruce Hornsby. Bigham wrote for and played percussion with Miles Davis, and Bigham's song "Jilli" appears on *Amandla* (Warner Brothers, 1989), Davis's last studio album. Bigham also appears on the DVD *Miles Davis-Live in Paris* (Warner Brothers, 2001).

TOYIA BROWN

*The Good Girl Blues*/John "JB" Bigham

*The Good Girl Blues* was recorded in Bigham's home studio—Whitley Manor—in the heart of Hollywood's historic Whitley Heights area. "Actually, the studio was in my closet initially, but I just graduated to the second bedroom," he exclaims. "I caught my girlfriend with her guard down! Once I set it up, she saw how cool it was."

Not one to amass gear just for the sake of it, Bigham's modest setup suits him well. He records into an Apple Power Mac G4 (running Digidesign Pro Tools LE and Propellerhead Reason) through a Digi 001. He says that his first recording rig was an Atari ST, after which he graduated to a Roland VS-1680, and then to his Mac.

### RIFFS

**The Soul of John Black**

**Home base:** Los Angeles, California
**Key software:** Digidesign Pro Tools LE and Reason 3.04
**Mics of choice:** Shure SM48 and SM57
**Web site:** www.thesoulofjohnblack.com

Bigham owns only a couple of mics. "I'm not inspired to buy much new stuff; I don't have the money. I'm a big fan of Jon Brion, and he said that if you don't have a lot of money, get one mic you like and use it on everything. I actually have two: a Shure SM48 I got 15 years ago and an SM57 that I mic my amplifiers and record my guitar with. I have a Neumann TLM170, but that's borrowed. For me, it starts with the instrument sounding good. Whether you record low-fi or hi-fi, it's going to come through because the essence is good. When you listen to that old blues stuff, you feel the energy. It's not like they had a brand-new Gibson; they had whatever they had, and it was captured. The only people who care about technical stuff are technical people. This system works great for me."

Bigham played all the guitar and Clavinet on the CD, and he programmed the drums. "I did all the drums in Reason 3.0.4. I have a dbx 160× compressor/limiter and mic pres by the Mastering Lab in Hollywood and M-Audio (the Audio Buddy). I wanted it to feel it was played by a real drummer. My favorite thing to use is Reason's Rex Loops. That's the mustard for me. It's over the top. I like to keep my drum parts minimal, but then I'll add these rhythmic loops."

One of the highlights of the CD is the gospel-tinged background vocals provided by the talented trio of Laura Jane Jones (who has sung with Enrique Iglesias), JoNell Kennedy (who played Joann in *Dreamgirls*), and Kandace Linsey (who sings with Marc Anthony).

JB's homage to the blues artists that have influenced him is evident on *The Good Girl Blues*. "John Lee Hooker said, 'Real blues—that's what you call soul. And I'm gonna stay in Soulville a little while,'" recalls Bigham, who then adds, "Me too!" EM

# alphatrack

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# Digital Performer scores **the impossible**

*"Digital Performer is a huge part of my scoring process. I do all of my writing in DP. At the Mission Impossible 3 sessions, we had a laptop running DP and a MOTU Traveler to handle prelays and record live stereo stems of Dan Wallin's mix from the main board. DP also drove video to the main monitors and synced the entire 100+ piece orchestra, so that everything was perfectly in line with my composition sequence. I count on DP every day. It performs flawlessly."*

— ***Michael Giacchino***
*Composer*
*Original Music for M:i:3*

*Mission Impossible 3 "Bridge Battle" Digital Performer project courtesy of Michael Giacchino and Chad Seiter. Mission Impossible 3 image courtesy of Paramount Pictures. All rights reserved.*


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